Patent
Attorney's Docket No. RIC95042

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of:	) **Mail Stop Amendment**
ALEXANDER LISHENG HUANG	) Group Art Unit: 2616
Application No.: 08/575,433	) Examiner: P. Tran
Filed: December 20, 1995	)
For: HYBRID PACKET-SWITCHED AND CIRCUIT-SWITCHED TELEPHONY SYSTEM	)

DECLARATION UNDER 37 C.F.R. § 1.131

U.S. Patent and Trademark Office
Customer Window, Mail Stop Amendment
Randolph Building
401 Dulany Street
Alexandria, VA 22314

Sir:

I, Alexander Lisheng Huang, hereby declare that:

1. I am the sole inventor of the invention disclosed and claimed in U.S. Patent Application No. 08/575,433, filed December 20, 1995 ("the present application"), and I conceived of this invention prior to September 20, 1995. Since the date of my original patent application, I have legally changed my name from Lisheng Huang to Alexander Lisheng Huang, and my application has been changed to reflect this name change.

2. A "Disclosure of Invention" document related to the present application and in which I set forth a "Description of Invention," was signed by me and witnessed prior to September 20, 1995. **Exhibit A** includes a redacted copy of the "Disclosure of Invention" document.

8 OCT 2007, 1/4

3. A communication was sent to me from Pollock, Vande Sande & Priddy, RLLP ("our attorneys") prior to September 20, 1995, that referenced their receipt of the Disclosure of Invention, which was assigned Docket No. RIC-95-042. The communication requested additional information regarding my invention. **Exhibit B** includes a redacted copy of the communication.

4. A communication was sent to me from our attorneys prior to September 20, 1995, which provided three documents for me to consider in relation to my invention. **Exhibit C** includes a redacted copy of the communication.

5. An electronic mail message and subsequent facsimile were sent to our attorneys from me prior to September 20, 1995, which included additional information regarding my invention. **Exhibit D** includes a redacted copy of the electronic mail message and subsequent facsimile.

6. A communication was sent to me from our attorneys prior to September 20, 1995, that provided search result documents for me to consider in relation to my invention. **Exhibit E** includes a redacted copy of the communication.

7. A first draft of the present application was completed prior to September 20, 1995 by our attorneys. A letter accompanying and referencing the completed first draft patent application was written by our attorneys and sent to me at MCI Communications Corporation. **Exhibit F** includes a redacted copy of the letter.

8. An electronic mail message was sent to our attorneys from me prior to September 20, 1995, which included comments regarding the search result documents. **Exhibit G** includes a redacted copy of the electronic mail message.

8OCT2001, 2/4

9. An electronic mail message was sent to our attorneys from me prior to September 20, 1995 that requested an electronic copy of the first draft of the patent application. **Exhibit H** includes a redacted copy of the electronic mail message.

10. An electronic mail message was sent to me from our attorneys prior to September 20, 1995 that provided an electronic copy of the first draft patent application. **Exhibit I** includes a redacted copy of the electronic mail message.

11. An electronic mail message was sent to our attorneys from me prior to September 20, 1995 that included my comments on the first draft patent application. **Exhibit J** includes a redacted copy of the electronic mail message.

12. I further declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and, further, that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements made jeopardize the validity of the application or any patent issuing therefrom.

Date: 8 OCT. 2007

Signature: ______________
Alexander Lisheng Huang

U.S. Patent Application No. 08/575,433
Attorney's Docket No. RIC95042

LIST OF EXHIBITS

A: A copy of an Invention Disclosure, describing a portion of the claimed subject matter, signed and witnessed prior to September 20, 1995.

B: A copy of a letter sent to MCI Communications Corporation, dated prior to September 20, 1995, which referenced receipt of the Invention Disclosure and requested additional information regarding the invention.

C: A copy of a communication sent to MCI Communications Corporation, dated prior to September 20, 1995, which provided three documents for consideration in relation to the invention.

D: A copy of an electronic mail message and subsequent facsimile sent to our attorneys, dated prior to September 20, 1995, which included additional information regarding the invention.

E: A copy of a communication sent to MCI Communications Corporation, dated prior to September 20, 1995, which provided search result documents.

F: A copy of a letter sent to MCI Communications Corporation, dated prior to September 20, 1995, which accompanied and referenced a first draft application.

G: A copy of an electronic mail message sent to our attorneys, dated prior to September 20, 1995, which included comments on the search result documents.

H: A copy of an electronic mail message sent to our attorneys, dated prior to September 20, 1995, which requested an electronic copy of the first draft application.

I: A copy of an electronic mail message sent to MCI Communications Corporation, dated prior to September 20, 1995, which included an electronic copy of the first draft application.

J. A copy of an electronic mail message sent to our attorneys, dated prior to September 20, 1995, which included comments on the first draft application.

8 OCT 2007 4/4

EXHIBIT A

DISCLOSURE OF INVENTION

Descriptive Title: HYBRID PACKET-SWITCHED AND CIRCUIT-SWITCHED TELEPHONY

Inventor Names: Huang, Lisheng (a.k.a. "Alex") 214-918-4912
Last First Middle Tel.

Existing Documentation: None.

Problem To Be Solved:

(1) High transmission and access costs of circuit-switched telephony, and

(2) Customer premises equipment investment and inconvenience of pure packet-switched telephony such as Internet phones.

Summary of Invention:

A hybrid packet and circuit switched telephony system routes a telephone call mostly through packet-switched networks, except for the two ends where the telephone sets of the caller and the callee are directly connected. The key element of system is a Gateway Computer (GC) which converts voice signals into data packets and vice versa, resolves the destinations, and routes the packets. GCs are managed by the telephony service provider, as opposed to the end-user.

The benefits of this hybrid packet and circuit switched telephony are:

(1) lower cost of transport due to better utilization of packet-switched network over pure circuit-switched network;

(2) availability to any consumer at no initial investment as required by pure packet-switched telephony, such as purchasing a multimedia personal computer or Internet access; and

(3) the potential for unlimited intelligent services, due to computer based telephony, such as caller's personalized speed dialing list, callee personalized virtual destination number, and integration with electronic mails, etc.

Known Prior Art:

Himowitz on computers: Phoning on Internet may be next revolution, 3/3/95, Los Angeles Times-Washington Post.

Read and Understood:

(Witness) [redacted] (Date) Signature(s) of Inventors [redacted] (Date)

(Witness) [redacted] (Date)

[redacted]

Description of Invention:

GCs are deployed to connect both packet-switched networks, such as the Internet, and circuit-switched networks, such as LECs, as demonstrated by elements 3 and 6 in Figure 1.

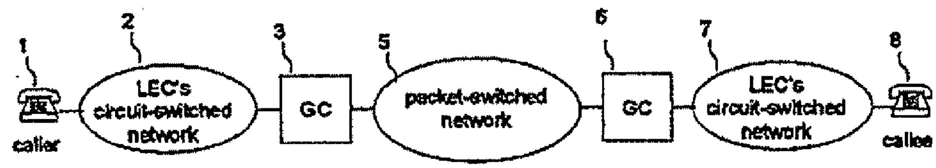

Figure 1.

The GC shall have the functions:

(a) to encode/decode voice signal into/from data,

(b) to compress/decompress the data for efficiency,

(c) to optionally encrypt/decrypt the data for security,

(d) to packet/unpack the data for transmission in the packet-switched network (5 in Figure 1.),

(e) to resolve the destination of a call and the terminating GC,

(f) to interpret control requests by the caller in the forms of telephone keypad tones (tone reception) or spoken commands (speech recognition),

(g) to connect to both circuit-switched networks (out-of-band signaling preferred) and packet switched networks, and

(h) to support protocols among GCs to locate the terminating GC to serve the callee, as well as other handshaking implied by the scenarios given below.

The steps of a hybrid packet and circuit switched call can be shown in Figure 2.

Read and Understood:

(Witness)	[redacted] (Date)	Signature(s) of Inventors	[redacted] (Date)
(Witness)	[redacted] (Date)		

[redacted] [redacted] [redacted]

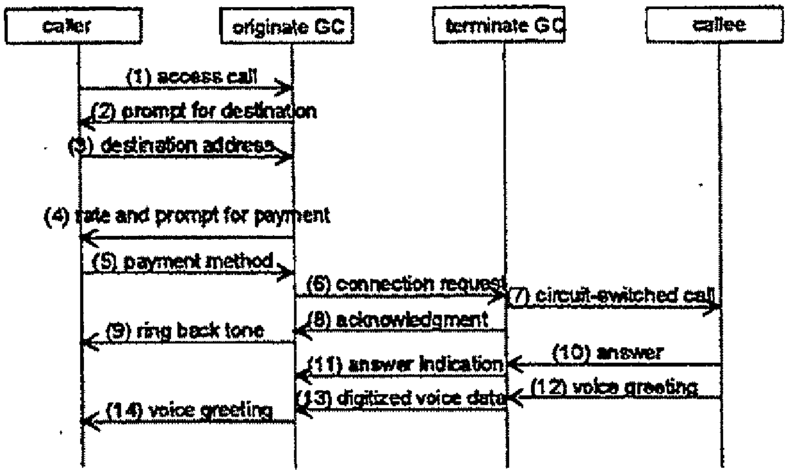

Figure 2.

The descriptions of the Figure 2 are as follows:

(1) The caller first calls a local GC ("originate GC") through a Local Exchange Carrier (LEC) from any telephone; the caller's address (caller's telephone number) is usually passed to the GC by the LEC;

(2) The originate GC plays a voice prompt (a greeting message asking for input) to the caller for the callee's destination address (callee's telephone number).

(3) The caller provides the address either through telephone keypad digits or through spoken digits which are recognized by the GC;

(not shown) The originate GC resolves the address similar to the Domain Name Service for the Internet and obtains the packet network address (such as the IP address of the Internet) of the terminate GC, which is usually local to the callee (otherwise a toll call may be involved). Meanwhile, it estimates the unit charge for a call going through that terminate GC;

(4) The originate GC informs the caller about the charge rate, and asks for caller's preferred payment method, such as by credit card, or through an existing user account.

(5) The caller specifies the payment method either through keypad digits or through spoken digits which are recognized by the GC; If this is a collect call, then the caller's spoken information about both parties are recorded and digitized to be announced later to the callee (scenario not shown);

(not shown) The originate GC validates the payment method through internal or external databases;

(6) The originate GC sends a control message the terminate GC, along with both parties addresses; if the terminate GC does not know where to route the call or does not have the resource to serve the call, it responds with a negative acknowledgment and an alternate terminate GC is searched for, or the caller is so informed (not shown);

Read and Understood:

(Witness) [redacted] (Date) Signature(s) of Inventors [redacted] (Date)

(Witness) [redacted] (Date)

(7) The terminate GC dials out to the callee through his/her LEC.

(8) If the call proceeds successfully through the LEC, the terminate GC sends an acknowledgment back to the originate GC (the unsuccessful, mostly busy, scenario is not shown);

(9) The originate GC then passes the status back to the caller through the LEC, the effect being a ring back tone;

(10) The callee answers the call;

(11) The terminate GC passes this state change to the originate GC which may start billing;

(12) The callee starts the conversation by greeting the caller;

(13) The terminate GC continuously digitizes all the signals from the callee, possibly encrypts, compresses, and packs into packets the data, and sends the packets over the network to the originate GC;

(14) The originate GC continuously unpacks, decompresses, and possibly decrypts, the data, and converts the data back to voice to the caller over the LEC;

(not shown) The same process is performed for the caller's voice, in the opposite direction of the one described in steps 12 through 14; the processing in both directions supports the conversation between the two parties in the call.

The above networking is symmetric. Sometimes a caller may have a multimedia-capable computer and a packet network connection. Using this kind of customer premises equipment, instead of a plain telephone set, may enable the caller for advanced services or features. However, the callee does not have (a) a multimedia computer, or (b) a packet network access, or (c) the same packet-switched telephony application currently running. Therefore a GC support, based on the same technique but an asymmetric configuration, is useful, as shown in Figure 3.:

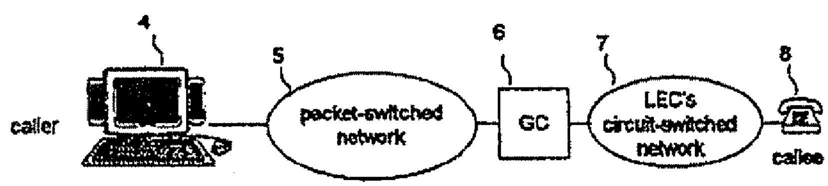

Figure 3.

The caller's multimedia computer (element 4 in Figure 3.) actually runs an application compatible to a GC's protocol and acts almost as the originate GC from the terminate GC's view point, except the billing and validation of the caller may be performed by the terminate GC (element 6 in Figure 3.), possibly with the assistance of caller's access point to the packet network.

Read and Understood:

(Witness) [redacted] (Date) Signature(s) of Inventors [redacted] (Date)

(Witness) [redacted] (Date)

[redacted] [redacted] [redacted]

EXHIBIT B

Date: [redacted] EST
From: Townsend Belser / MCI ID: 729-0130

TO: Alex Huang / MCI ID: 727-8033
CC: Denise Nappi / MCI ID: 746-5183
Subject: RIC-95-042
Message-Id: 34950504144843/0007290130PJ3EM

As you know, we have been asked by the Office of the General Counsel (OGC) to protect a valuable asset of MCI, namely your invention, by preparing and filing a patent application of same. Please contact either Marianne Geeker or Savery Gradoville of OGC for confirmation or if you need more detail.

It is my understanding that John Hoel of our firm had previously contacted you (or someone who works with you who informed him that you are the inventor) and that you had provided him a very preliminary disclosure of your invention.

After having reviewed the preliminary disclosure, to further proceed with the process of preparing the application, we now need to ask you to provide us with the following additional information:

A. FIGURES.

(We believe the following figures are necessary to the understanding of your invention insofar as your invention appears to relate to software.)

1. A process flow diagram showing the steps performed when your invention is operating.

2. A system block diagram showing how the parts of your invention and the system with which it interacts, are connected together.

3. A "before and after" diagram showing the effect caused by operating your invention. Some examples of this type of diagram are:

   a data table shown before and after the invention acts

   a display screen shown before and after the invention acts

   a routing path shown before and after the invention acts

   a waveform shown before and after the invention acts

B. TEXT DESCRIBING THE OPERATION OF YOUR INVENTION.

This can be a narrative description of where the steps in your process flow diagram (Figure A.1. above) are performed

in your system block diagram (Figure A.2 above) to create the change shown in your "before and after" diagram (Figure A.3. above).

C. STATE THE PROBLEM THAT IS SOLVED BY YOUR INVENTION.

D. BACKGROUND OF THE INVENTION

This may be considered as a subparagraph to C inasmuch as the problem to be solved and the background are usually related.

To the extent that you have any information, please identify the efforts of others which have failed to solve the problem that your invention solves.

Do note that neither the figures nor the text need be especially created for this invention disclosure. To the extent the requested information already exists, in whatever form, that existing material can be used, as long as it provides the necessary information.

Since we have asked our paralegal Denise Nappi to manage the various dockets, please send your additional disclosure to her, either by MCI Mail or fax. (Denise can be reached by MCI Mail.) If you are sending the additional disclosure by FAX, please label your FAX with your invention disclosure number and send it to (202) 293-6229.

If you want to reply by Federal Express or regular mail, please direct your mail to our address below and address it to Denise Nappi's attention.

Please include your invention disclosure number on all correspondence.

Pollock, Vande Sande and Priddy
1990 M Street NW, Suite 800
Washington DC 20036
Phone (202) 331-7111
FAX (202) 293-6229

In the meantime, if you have any questions relating to the above or about your invention, or general questions relating to patents or how best to proceed, please do not hesitate to call either me or John Hoel per the above-noted phone number.

Thanks,

Townsend Belser
\
\

EXHIBIT C

LAW OFFICES
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1990 M STREET, N. W.
WASHINGTON, D. C. 20036-3425

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GEORGE VANDE SANDE
ROBERT R. PRIDDY
RICHARD WIENER
BURTON A. AMERNICK
STANLEY B. GREEN
MORRIS LISS
TOWNSEND M. BELSER, JR.
MARTIN ABRAMSON
GEORGE R. PETTIT
THOMAS J. VANDE SANDE
LOUIS WOO
ELZBIETA CHLOPECKA
ERIC J. FRANKLIN
LARRY LIBERCHUK**

OF COUNSEL
JOHN E. HOEL*
JOSEPH C. REDMOND, JR.
*MEMBER VIRGINIA BAR ONLY
**MEMBER PENNSYLVANIA BAR ONLY

INTELLECTUAL PROPERTY CAUSES

TELEPHONE (202) 331-7111

FAX 202-293-6229
FAX 202-223-2596

P.O. BOX 19088
WASHINGTON, D.C. 20036-0088

[redacted]

VIA FEDERAL EXPRESS

Mr. Lisheng "Alex" Huang
MCI Communications Corporation
2400 N. Glenville Drive
Richardson, Texas 75082

Re: Invention Disclosure on
HYBRID PACKET-SWITCHED AND
CIRCUIT-SWITCHED TELEPHONY
Your Ref: RIC-95-042
Our Ref: 1643/339

Dear Alex:

I am assisting John Hoel in the processing of your above invention disclosure which we received by fax from Jay Shah on [redacted].

Enclosed are two articles by David J. Goodman, namely: (1) pages 31-40 with "Trends In Cellular And Cordless Communications", June 1991, IEEE Communications Magazine; and (2) pages 1272-1280 with "Cellular Packet Communications", August 1990, IEEE Transactions on Communications, Vol. 38, No. 8. Also enclosed is a copy of U.S. Patent No. 4,866,704 issued September 12, 1989, to Bergman.

In Fig. 5 of article (1), there is shown a cellular packet switch wherein two metropolitan area networks are connected together via Gateway Interface Units (GIU's), and a metropolitan area network includes Trunk Interface Units (TIU's) for connecting a metropolitan area network to a Public Switched Telecommunications Network (PSTN) and to a Broad Band Integrated Services Digital Network (BISDN).

Figs. 3 and 4 of article (2) show block diagrams of a cellular trunk interface unit and a wireless terminal unit, respectively. It would seem that the GIU of article (1) may have at least some features similar to those of Figs. 3 and 4 of article (2). It also appears that at least some of the call set-

up scenario of Fig. 7 in article (2) may be similar to the call set-up scenario described in your disclosure, except for the charging and payment validation steps.

Patent No. 4,866,704 describes a fiber optic voice/data network that supports ordinary data packet traffic simultaneously with synchronous T1 voice traffic over a common token ring channel, and transmits synchronous voice traffic over an asynchronous packet switched network in a way that creates a virtual T1 channel. Although the call set-up is not described in this reference, it appears that the system described might be capable of a call set-up scenario similar to that described in your disclosure.

I would appreciate your reviewing the enclosed references as soon as practicable, and then calling me to discuss how to best distinguish your invention from the disclosures of these references. This will help us prepare a patent application on your disclosure.

Please confirm by MCI Mail your receipt of the original of this letter and its enclosures and then call me as soon as you have had an opportunity to consider the enclosures.

I look forward to hearing from you soon. Please also call if you have any questions.

With my best regards,

Cordially,

POLLOCK, VANDE SANDE & PRIDDY

BY: Townsend M. Belser, Jr.

TMB/cdc
Enclosures

Advance Copy of Letter Only VIA MCI Mail

cc: Mr. Jay Shah (letter only) via MCI Mail

EXHIBIT D

TMB, [redacted]
I talked to Mr.
Huang & he
said he would
pay this.
Denise

**RECEIPT

Date: [redacted] EST
From: Alex Huang / MCI ID: 727-8033

TO: * Denise Nappi / MCI ID: 746-5183
CC: Townsend Belser / MCI ID: 729-0130
CC: Alex Huang / MCI ID: 727-8033
Subject: Re: RIC-95-042

Dear Denise,

The enclosed is a Microsoft Word 6.0 document in response to the
memo
below. If you cannot retrieve it due to software problem, or if
you
need more information, please let me know. I am sorry for this
late
response.

Thank you for your help.

Alex Huang
(214)918-4912

> Date: [redacted] CDT
> From: Townsend Belser / MCI ID: 729-0130
>
> TO: * Alex Huang / MCI ID: 727-8033
> CC: Denise Nappi / MCI ID: 746-5183
> Subject: RIC-95-042
>
> As you know, we have been asked by the Office of the General
>
> Counsel (OGC) to protect a valuable asset of MCI, namely your
>
> invention, by preparing and filing a patent application of same.
>
> Please contact either Marianne Geeker or Savery Gradoville of OGC

>
> for confirmation or if you need more detail.
>
>
>
> It is my understanding that John Hoel of our firm had previously
>
> contacted you (or someone who works with you who informed him
>
> that you are the inventor) and that you had provided him a very
>
> preliminary disclosure of your invention.
>
>
>

Facsimile Cover Sheet for
MCI's Network Engineering Lab

To: Denise Nappi
Company:
Phone:
Fax:

From: Lisheng "Alex" Huang
Company: MCI
Phone: 214-918-4912
Fax: 214-907-8784

Date:
Pages include cover sheet: 9
(Please contact sender if you do not recieve all pages of this fax)

Comments: Your Ref.: 1643/339
Our Ref.: RIC-95-042

INVENTOR: Lisheng ("Alex") Huang

TITLE: Hybrid Packet-Switched And Circuit-Switched Telephony
Ref: RIC-95-042
(POLLOCK, VANDE SANDE & PRIDDY's Ref: 1643/339)

PROBLEM SOLVED: (1) High transmission and access costs of circuit-switched telephony, and (2) Customer premises equipment investment and inconvenience of pure packet-switched telephony such as Internet phones.

A. FIGURES

Fig. 1a - Flow Diagram of Process ("GC" stands for "Gateway Computer")

(1) The caller first calls a local GC ("originate GC") through a Local Exchange Carrier (LEC) from any telephone; the caller's address (caller's telephone number) is usually passed to the GC by the LEC;

(2) The originate GC plays a voice prompt (a greeting message asking for input) to the caller for the callee's destination address (callee's telephone number).

(3) The caller provides the address either through telephone keypad digits or through spoken digits which are recognized by the GC;

(4) The originate GC resolves the address similar to the Domain Name Service for the Internet and obtains the packet network address (such as the IP address of the Internet) of the terminate GC, which is usually local to the callee (otherwise a toll call may be involved). Meanwhile, it estimates the unit charge for a call going through that terminate GC;

(5) The originate GC informs the caller about the charge rate, and asks for caller's preferred payment method, such as by credit card, or through an existing user account.

(6) The caller specifies the payment method either through keypad digits or through spoken digits which are recognized by the GC; If this is a collect call, then the caller's spoken information about both parties are recorded and digitized to be announced later to the callee (scenario not shown);

(7) The originate GC validates the payment method through internal or external databases;

(8) The originate GC sends a control message the terminate GC, along with both parties addresses; if the terminate GC does not know where to route the call or does not have the resource to serve the call, it responds with a negative acknowledgment and an alternate terminate GC is searched for, or the caller is so informed (not shown);

(9) The terminate GC dials out to the callee through his/her LEC.

(10) If the call proceeds successfully through the LEC, the terminate GC sends an acknowledgment back to the originate GC (the unsuccessful, mostly busy, scenario is not shown);

(11) The originate GC then passes the status back to the caller through the LEC, the effect being a ring back tone;

(12) The callee answers the call;

(13) The terminate GC passes this state change to the originate GC which may start billing;

(14) The callee starts the conversation by greeting the caller;

(15) The terminate GC continuously digitizes all the signals from the callee, possibly encrypts, compresses, and packs into packets the data, and sends the packets over the network to the originate GC;

(16) The originate GC continuously unpacks, decompresses, and possibly decrypts, the data, and converts the data back to voice to the caller over the LEC;

(17) The same process is performed for the caller's voice, in the opposite direction of the one described in steps 12 through 14; the processing in both directions supports the conversation between the two parties in the call.

Fig. 1b - Flow Diagram of Process for primary service

(1) The caller makes a long-distance call by dialing a destination address (callee's telephone number) through a Local Exchange Carrier (LEC) from his/her dedicated telephone such as home phone or office phone for which a routing configuration to a Gateway Computer (GC) is preset with the LEC. The LEC routes the call to the GC. The caller's address (caller's telephone number) is usually passed to the GC by the LEC along with the destination address.

↓

(2) The originate GC authorizes the call by checking the caller's account information through internal databases; It also resolves the routing using the dialed destination address.

↓

(3) The originate GC then sends a control message the terminate GC, along with both parties addresses; if the terminate GC does not know where to route the call or does not have the resource to serve the call, it responds with a negative acknowledgment and an alternate terminate GC is searched for, or the caller is so informed.

↓

(4) The terminate GC dials out to the callee through his/her LEC.

↓

(5) If the call proceeds successfully through the LEC, the terminate GC sends an acknowledgment back to the originate GC (the unsuccessful, mostly busy, scenario is not shown);

↓

(6) The originate GC then passes the status back to the caller through the LEC, the effect being a ring back tone;

↓

(7) The callee answers the call;

↓

(8) The terminate GC passes this state change to the originate GC which may start billing;

↓

(9) The callee starts the conversation by greeting the caller;

↓

(10) The terminate GC continuously digitizes all the signals from the callee, possibly encrypts, compresses, and packs into packets the data, and sends the packets over the network to the originate GC;

↓

(11) The originate GC continuously unpacks, decompresses, and possibly decrypts, the data, and converts the data back to voice to the caller over the LEC;

↓

(12) The same process is performed for the caller's voice, in the opposite direction of the one described in steps 9 through 11; the processing in both directions supports the conversation between the two parties in the call.

Fig. 2 - System Block Diagram

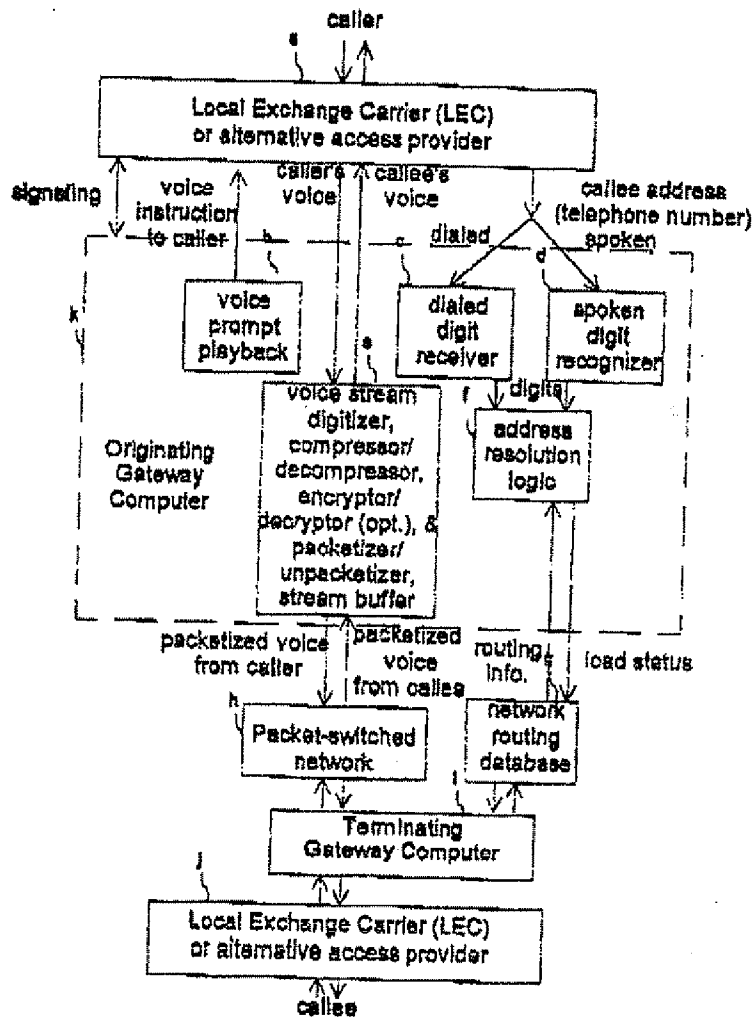
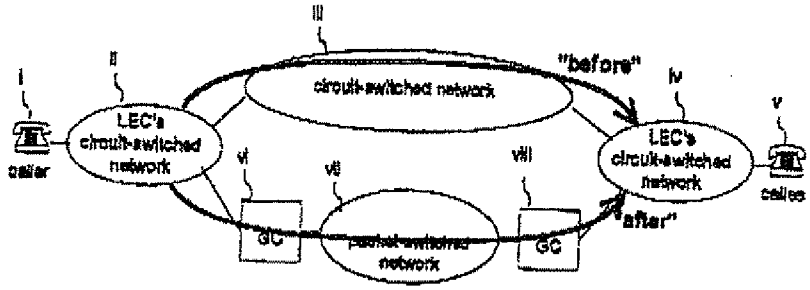
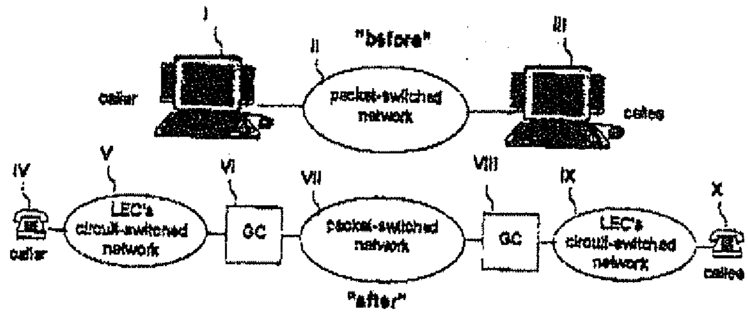


Fig. 3 - "Before and After" Diagram



3a



3b

B. TEXT DESCRIBING THE OPERATION OF THE INVENTION

In the context of this document, the invention, "Hybrid Packet-Switched And Circuit-Switched Telephony" is abbreviated as "HPCT". It should be noted that the emerging Asynchronous Transfer Mode (ATM) networking is treated in this context as a special case of packet-switched networking with low delay and low delay deviation, although they are often categorized as a third type of networking -- "cell-switched networking".

The caller can use the HPCT service as an alternative long-distance-telephony service ("alternative service"), or he/she can use this as his/her primary long-distance-telephony service ("primary service"). Alternative service can be used from any telephone while primary service can be used only from subscriber's dedicated telephone, such as home phone or office phone. "Alternative service" may also be called "casual service" where the caller does not have to have an account preset with the service provider; rather, the authorization is through a credit card service.

Fig. 1a shows the alternative service case:

(1) The caller first calls a local Gateway Computer ("originate GC") through a Local Exchange Carrier (LEC) from any telephone; the caller's address (caller's telephone number) is usually passed to the GC by the LEC. This is also reflected by information flow from "caller" to Block a, and then to Block k, in Fig. 2. In Fig. 3, this step corresponds to a route from Block i to Block ii and then to Block iv, or route from Block IV to Block V and then to Block VI.

(2) The originate GC plays a voice prompt (a greeting message asking for input) to the caller for the callee's destination address (callee's telephone number). This is also reflected by information flow from Block b to Block a, and then to "caller", in Fig. 2.

(3) The caller provides the address either through telephone keypad digits or through spoken digits which are recognized by the GC). This is also reflected by information flow from "caller" to Block c or Block d in Fig. 2.

(4) The originate GC resolves the address similar to the Domain Name Service for the Internet and obtains the packet network address (such as the IP address of the Internet) of the terminate GC, which is usually local to the callee (otherwise a toll call may be involved). Meanwhile, it estimates the unit charge for a call going through that terminate GC. This is also reflected by information flow from Block c or Block d to Block f, and in-between Blocks f and g, in Fig. 2.

(5) The originate GC informs the caller about the charge rate, and asks for caller's preferred payment method, such as by credit card, or through an existing user account. This is also reflected by information flow from Block b to Block a, and then to "caller", in Fig. 2.

(6) The caller specifies the payment method either through keypad digits or through spoken digits which are recognized by the GC (If this is a collect call, then the caller's spoken information about both parties are recorded and digitized to be announced later to the callee).

(7) The originate GC validates the payment method through internal or external databases (not shown in Fig. 2).

(8) The originate GC sends a control message the terminate GC, along with both parties addresses; if the terminate GC does not know where to route the call or does not have the resource to serve the call, it responds with a negative acknowledgment and an alternate terminate GC is searched for, or the caller is so informed. This is also reflected by information flow from Block k to Block h, and then to Block i, in Fig. 2. In Fig. 3, this step corresponds to a route from Block vi to Block vii and then to Block viii, or route from Block VI to Block VII and then to Block VIII.

(9) The terminate GC dials out to the callee through his/her LEC. This is also reflected by information flow from Block i to Block j, and then to "callee", in Fig. 2. In Fig. 3, this step corresponds to a route from Block viii to Block iv and then to Block v, or route from Block VIII to Block IX and then to Block X.

(10) If the call proceeds successfully through the LEC, the terminate GC sends an acknowledgment back to the originate GC (the unsuccessful, mostly busy, scenario is not shown);

(11) The originate GC then passes the status back to the caller through the LEC, the effect being a ring back tone;

(12) The callee answers the call;

(13) The terminate GC passes this state change to the originate GC which may start billing;

(14) The callee starts the conversation by greeting the caller;

(15) The terminate GC continuously digitizes all the signals from the callee, possibly encrypts, compresses, and packs into packets the data, and sends the packets over the network to the originate GC. This is also reflected by information flow from "callee" to Block j, Block i, Block h, and then to Block e, in Fig. 2.

(16) The originate GC continuously unpacks, decompresses, and possibly decrypts, the data, and converts the data back to voice to the caller over the LEC. This is also reflected by information flow from Block e, to Block a, and then to the "caller", in Fig. 2.

(17) The same process is performed for the caller's voice, in the opposite direction of the one described in steps 12 through 14; the processing in both directions supports the conversation between the two parties in the call. This is also reflected by information flow from "caller" to Block a, Block e, Block h, Block i, Block j, and then to the "callee", in Fig. 2.

Fig. 1b shows the primary service case:

(1) The caller makes a long-distance call by dialing a destination address (callee's telephone number) through a Local Exchange Carrier (LEC) from his/her dedicated telephone such as home phone or office phone for which a routing configuration to a Gateway Computer (GC) is preset with the LEC. The LEC routes the call to the GC. The caller's address (caller's telephone number) is usually passed to the GC by the LEC along with the destination address. This is also reflected by information flow from "caller" to Block a, and then to Block k through the signaling link, in Fig. 2.

(2) The originate GC authorizes the call by checking the caller's account information through internal databases; It also resolves the routing using the dialed destination address. This is also reflected by information flow from Block k to Block f, and then to Block g, in Fig. 2.

(3) The originate GC then sends a control message the terminate GC, along with both parties addresses; if the terminate GC does not know where to route the call or does not have the resource to serve the call, it responds with a negative acknowledgment and an alternate terminate GC is searched for, or the caller is so informed. This is also reflected by information flow from Block k to Block h, and then to Block i, in Fig. 2.

(4) The terminate GC dials out to the callee through his/her LEC. This is also reflected by information flow from Block i to Block j, and then to the "callee", in Fig. 2. In Fig. 3, this step corresponds to a route from Block viii to Block iv and then to Block v, or route from Block VIII to Block IX and then to Block X.

(5) If the call proceeds successfully through the LEC, the terminate GC sends an acknowledgment back to the originate GC (the unsuccessful, mostly busy, scenario is not shown);

(6) The originate GC then passes the status back to the caller through the LEC, the effect being a ring back tone;

(7) The callee answers the call;

(8) The terminate GC passes this state change to the originate GC which may start billing;

(9) The callee starts the conversation by greeting the caller;

(10) The terminate GC continuously digitizes all the signals from the callee, possibly encrypts, compresses, and packs into packets the data, and sends the packets over the network to the originate GC. This is also reflected by information flow from "callee" to Block j, Block i, Block h, and then to Block e, in Fig. 2.

(11) The originate GC continuously unpacks, decompresses, and possibly decrypts, the data, and converts the data back to voice to the caller over the LEC. This is also reflected by information flow from Block e, to Block a, and then to the "caller", in Fig. 2.

(12) The same process is performed for the caller's voice, in the opposite direction of the one described in steps 9 through 11; the processing in both directions supports the conversation between the two parties in the call. This is also reflected by information flow from "caller" to Block a, Block e, Block h, Block i, Block j, and then to the "callee", in Fig. 2.

The key sub-system of HPCT is Block e in Fig. 2. It has many important functions, including real-time compression of digitized voice and stream buffering.

The expected compression ratio may achieve 25:1 or better with the emerging technology, and the consideration of silence in the speech may double that ratio. This can make the HPCT very efficient and very cost-effective. The compression at this kind of high ratio are likely to be information-lossy, but it is expected to be both tolerated by the human ear and compensated by the low price of the service.

The buffering mechanism at the receiving end can mostly recover the stream of voice from packets arriving with variable delay introduced by the packet network.

C. BACKGROUND AND PROBLEM SOLVED BY THE INVENTION

There are currently emerging software products to use Internet, which is a set of interconnected packet-switched networks, for telephony. VocalTec software provides half-duplexed long-distance telephone capability through Internet (Himowitz on computers: Phoning on Internet may be next revolution, 3/3/95, Los Angeles Times-Washington Post). Camelot Corp. is the latest entry in the Internet phone business with a $99.95 Mosaic front end that supports full-duplex voice conversation. (From PC Week for February 20, 1995 by Andy Patrizio). These products provide an alternative to long-distance telephone service for the users. They digitize and compress voice and transport over Internet. Some limitations of this pure packet networking are: (1) Both the caller and the callee must have computers (currently IBM PC compatibles), (2) they must have sound system on the computer, (3) they must have full Internet access, and (4) they must be both running the same software at the time when the call is made. The first three limitations translate into a considerable amount of money which discounts the economy of the approach to telephony. The last limitation means the calls have to be scheduled in advance in most situations which certainly cannot match the convenience of regular telephone calls. An additional problem with these software products is that the performance is restricted by the CPU resource on PC, both the quality of the sound and the delay time of the processing.

HPCT, on the other hand, utilizes the Gateway Computer which is a set of resource shared by many users and thus with much higher utilization in telephony than a personal computer. Optimization of the performance can be achieved by using Digital Signal Processors (DSPs) or Application Specific Integrated Circuits (ASICs). The users do not have to specially invest or schedule calls. In fact, as described in Fig. 1b the users may not tell any difference between using HPCT and using their regular long-distance services except a much lower cost.

The other background of the invention is the traditional circuit-switched telephony. Considering the 50:1 compression discussed above in Section B, the utilization of the circuits in circuit-switched telephony can be only 1/50 of the HPCT; in other words, the cost of it can be 50 times higher than HPCT. Another important problem with circuit-switched telephony is the fact of proprietary nature of the telephony switches which are the foundation of this telephony. Because the switch software development is only done by the manufacturers, the cost and delay in adding new services are often frustrating and prohibiting. HPCT, however, are based on general-purpose computers with open architecture, which can open the development up and bring very cost-effective new services in much shorter time frame.

EXHIBIT E

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[redacted]

VIA FEDERAL EXPRESS

Mr. Lisheng "Alex" Huang
MCI Communications Corporation
2400 N. Glenville Drive
Richardson, Texas 75082

Re: Invention Disclosure on
HYBRID PACKET-SWITCHED AND
CIRCUIT-SWITCHED TELEPHONY
Your Ref: RIC-95-042
Our Ref: 1643/339

Dear Alex:

This is to let you know the results of our search in the records of the U. S. Patent and Trademark Office to help evaluate the desirability of proceeding with a patent application on your above disclosure.

The search was directed to a hybrid packet-switched and circuit-switched telephony (HPCT) system for routing a telephone call mostly through packet-switched networks, except for respective analog ends where subscriber telephone sets are directly connected to circuit-switched networks. Gateway Computers (GC's) interconnect the packet-switched and circuit-switched networks to convert voice signals into data packets and vice versa, and to resolve the call destinations while routing the packets. In this invention, the GC's are managed by the telephony service provider, as opposed to the end-user. Because the GC's are a set of resources shared by many subscribers, they can be managed with higher efficiency and utilization than by a subscriber's personal computer. By incorporating the HPCT system into the current long-distance telephony, lower cost of communication and other packet-switch benefits can be achieved due to better utilization of packet-switched networks by the circuit-switched networks of local exchange carrier's (LEC's). These benefits are also made available to many subscribers without significant investment.

The following prior art patents were located by our search and copies thereof are enclosed for your consideration:

Patent No.	Issued	Inventor
4,549,291	10/22/85	Renoulin et al.
4,592,048	05/27/86	Beckner et al.
4,723,238	02/02/88	Isreal et al.
4,926,416	05/15/90	Weik
5,014,266	05/07/91	Bales et al.
5,058,111	10/15/91	Kihara et al.
5,119,370	06/02/92	Terry
5,251,206	10/05/93	Calvignac et al.
5,301,189	04/05/94	Schmidt et al.
5,341,374	08/23/94	Lewen et al.
5,341,418	08/23/94	Yoshida
5,349,640	09/20/94	Dunn et al.
5,392,402	02/21/95	Robrock

No. 5,392,402 was selected as describing a broadband intelligent telecommunications network (BIN) having a fast packet (ATM) switch and capable of using the resources of conventional circuit-switched telephone networks. As shown in Fig. 4, the Line Information Database (LIDB) of the circuit-switched telephone network is adapted to be included in the resource system 63 for call handling support in the BIN. The resource system 63 includes a database 63A in the form of a LIDB 150 which is directly connected to a ATM port 70P. The LIDB 150 and the LIDB 150B may be connected to circuit-switched network elements through SS7 and X.25 links (column 8, line 17, to column 9, line 60). The database support of circuit-switched telephone networks is thereby made available to fast-packet networks to facilitate fast-packet system call handling. "Further, substantial economic value can be realized in fast-packet networks by using the existing LIDB infrastructure of the circuit-switched telephone network" (column 9, lines 15-25).

No. 5,058,111 was selected as describing subscriber line interface circuits that are between analog telephones and packet switched networks, and are capable of reducing the signal-to-noise ratio (S/N ratio) both in the analog/digital and the digital/analog conversions. Voice signals are transmitted and received as analog signals via line terminal circuits which are connected to respective packet communication signal processing circuits. In the packet communication signal processing circuits, packet assembly and disassembly, voice signal processing, and line terminal circuit control are carried out in synchronism with the packet communication network. Packet data are transferred serially between the packet communication signal processing circuits and a switch interface circuit 32, which interfaces with a packet switched network. Operation of the

subscriber line interface circuit package 30 is controlled by a microprocessor 31.

No. 5,301,189 was selected as describing a telecommunication network having both an asynchronous transfer mode (ATM) exchanges and synchronous transfer mode (STM) exchanges, and wherein interfaces allocated to the ATM exchanges serve the purpose of connecting transmission lines carrying the circuit-switched messages of the STM exchanges. The purpose of the invention is to provide the advantages of telecommunication networks operating in the ATM mode to the older currently installed telecommunication networks operating in the STM mode.

No. 5,341,374 was selected as describing a local communication network for integrating telephone voice data with digital data from other application devices and providing distributed call processing for all such data. Where applicable, the analog-to-digital conversion and storage operations are performed by the telephones 58. Digital data from the telephone is sent to an adapter card 70 connected to a node coupling unit 51 on a token ring network 50. One or more of the node coupling units may be connected to a bridge for providing connection to a gateway for an external network, such as a telephone central office.

No. 4,549,291 was selected as showing a hybrid local communication network that operates both in circuit and packet modes. The interface units on the loop 2 are referred to as Cluster Control Units and include a microprocessor 417 as shown in Fig. 4 of this patent. Each cluster control unit may be connected to a plurality of terminals, such as telephones, teleprinters, data terminals, or teletext or videotext receiver sets.

No. 4,592,048 was selected as disclosing an integrated packet switching and circuit switching system comprising a number of switching modules each connected to a different plurality of user terminals. Each switching module includes a packet switching unit used both to provide packet-switched communication channels among the user terminals connected to that switching module, and to switch control information between the user terminals and the control unit to establish both circuit-switched calls and packet-switched calls.

No. 4,723,238 was selected as showing an interface circuit for interconnecting a packet switch system to a circuit switch system, and for enabling a uniform dialing plan to be utilized to establish intra-system or inter-system connections. Each terminal of the packet switched system is assigned a unique

terminal address similar to the extension number used for terminals of the circuit switched system.

No. 4,926,416 was selected as describing a hybrid packet switching network wherein the packets of particular connections, such as voice, are treated with priority over packets of other connections. In order to avoid delay jitter and loss of information, all packets are divided within an exchange into subpackets of equal length and distributed to subframes, and switching takes place on the basis of the subframes.

No. 5,014,266 was selected as describing a circuit switching system that responds to a switching protocol to set up multiple logical links, compressed voice calls and subrate data calls on one logical channel between packet switching networks, voice concentrators and data multiplexers.

No. 5,119,370 was selected as showing a switching node for an optical communications network wherein short message traffic and connection signaling messages for longer duration and higher capacity traffic are both handled by a packet message switch of relatively low capacity and rapid response, while the longer duration and higher capacity traffic is handled by a circuit switch controlled by the signalling messages conducted via the packet message switch.

No. 5,251,206 was selected as describing a hybrid packet and circuit switching system allowing the merger of packet and circuit traffic from a plurality of user interface modules on a TDM bus. The data transfers through the hybrid switch are performed in bursts for the packet switched traffic as well as for the circuit switched traffic. The hybrid switch resolves the contentions and grants the transfer on a burst time basis, giving to the circuit switched traffic higher priority than for the packet switched traffic (column 6, lines 34-50).

No. 5,341,418 was selected as describing a terminal adapter for providing access from analog signal equipment of the four-wire full duplex type to an Integrated Services Digital Network System (ISDN).

No. 5,349,640 was selected as describing a ROLM Company Digital Telephone having a coder/decoder 130 for converting analog voice information to digital form and vice versa, and a microprocessor 118 enabling an application operating on this computing device to control or monitor operation of the telephone.

We would appreciate your reviewing the enclosed patents and letting us know the specific features of your disclosure that

distinguish it from the disclosures of these patents. This will help us evaluate the likelihood of successfully prosecuting a patent application on your disclosure.

Please confirm your receipt of this letter by telefax or MCI MAIL, and then let us have your comments on the enclosed patents as soon as practicable.

I look forward to hearing from you soon. Please call if you have any questions.

Thank you for your help.

Cordially,

Townsend

Townsend Belser

TMB/cdc
Enclosures

Advance Copy of Letter Only VIA MCI MAIL

cc via MCI MAIL (Letter Only): Mr. Tim D. Casey
Mr. Alex Huang
Ms. Denise Nappi

EXHIBIT F

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*NOT ADMITTED IN THE DISTRICT OF COLUMBIA

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WASHINGTON, D.C. 20036-0088

[redacted]

VIA FEDERAL EXPRESS

Mr. Lisheng "Alex" Huang
MCI Communications Corporation
2400 N. Glenville Drive
Richardson, Texas 75082

Re: U.S. Patent Application for
HYBRID PACKET-SWITCHED AND
CIRCUIT-SWITCHED TELEPHONY
Your Ref: RIC-95-042
Our Ref: 1643/339

Dear Alex:

Enclosed for your review and approval is a draft patent application (23 pages of specification and 4 informal sheets of drawings) on your above invention disclosure.

Please go over the enclosed application and drawings carefully and let us know by telefax or MCI Mail of any changes needed to put it in shape for filing in the U.S. Patent and Trademark Office. Please do this within the next few days as MCI's management has asked us to file this application as soon as possible.

If any revisions are needed, your comments and suggestions can be written on the enclosed copy which then should be returned to us by telefax. In this regard, please insert any additional disclosure that may be useful in further distinguishing the prior art patents enclosed with my letter to you of [redacted].

The specification, which concludes with a series of numbered claims, should accurately describe and claim the invention. If, after reading the application, you have any questions about the accuracy of the specification or drawings, or whether the claims adequately define the invention, please raise those questions with me by calling me at (202) 331-7111.

The law requires that a patent application contain a detailed description of the best mode contemplated by the inventor(s) for carrying out the invention, and that the description be complete enough for a person skilled in this art to make and use the invention. Please let us know if you feel that there might be anything missing in this regard.

Your careful attention to each of the above matters will help us ensure that the patent application filed is in the best possible condition for examination by the Patent Office and for issuance as a patent if the application is allowed.

Please call me at the above telephone number if you have any questions regarding any of the above matters.

We also need to know your residence address and citizenship.

Thanking you for your assistance in this matter,

Cordially,

POLLOCK, VANDE SANDE & PRIDDY

BY: Townsend M. Belser, Jr.

TMB/cdc
Enclosure: Draft patent application

Copy of letter **VIA MCI MAIL** - Tim D. Casey
Alex Huang
Denise Nappi

EXHIBIT G

ate: [redacted] EST
rom: Alex Huang / MCI ID: 727-8033

'O: * Townsend Belser / MCI ID: 729-0130
:C: Tim D. Casey / MCI ID: 201-8157
:C: Denise Nappi / MCI ID: 746-5183
;ubject: Re: Rich. Ref. RIC-95-42, Your Ref. 1643/339
lessage-Id: 55950817221455/0007278033PJ4EM

)ear Townsend:

Thank you for the patent search which I was very impressed by.

3elow please find my comments to the patents, following your
original text describing each patent. I apologize for this delayed
reply.

Sincerely,

Alex Huang

> Date: [redacted] CDT **RECEIPT
> From: Townsend Belser / MCI ID: 729-0130
>
> TO: * Alex Huang / MCI ID: 727-8033
> CC: Tim D. Casey / MCI ID: 201-8157
> CC: Denise Nappi / MCI ID: 746-5183
> Subject: Rich. Ref. RIC-95-42, Our Ref. 1643/339
>
> [redacted]
>
>
> VIA FEDERAL EXPRESS
> Mr. Lisheng "Alex" Huang
> MCI Communications Corporation
> 2400 N. Glenville Drive
> Richardson, Texas 75082
>
> Re: Invention Disclosure on
> HYBRID PACKET-SWITCHED AND
> CIRCUIT-SWITCHED TELEPHONY
> Your Ref: RIC-95-042
> Our Ref: 1643/339

Dear Alex:

This is to let you know the results of our search in the records of the U. S. Patent and Trademark Office to help evaluate the desirability of proceeding with a patent application on your above disclosure.

The search was directed to a hybrid packet-switched and circuit-switched telephony (HPCT) system for routing a telephone call mostly through packet-switched networks, except for respective analog ends where subscriber telephone sets are directly connected to circuit-switched networks. Gateway Computers (GC's) interconnect the packet-switched and circuit-switched networks to convert voice signals into data packets and vice versa, and to resolve the call destinations while routing the packets. In this invention, the GC's are managed by the telephony service provider, as opposed to the end-user. Because the GC's are a set of resources shared by many subscribers, they can be managed with higher efficiency and utilization than by a subscriber's personal computer. By incorporating the HPCT system into the current long-distance telephony, lower cost of communication and other packet-switch benefits can be achieved due to better utilization of packet-switched networks by the circuit-switched networks of local exchange carrier's (LEC's). These benefits are also made available to many subscribers without significant investment.

The following prior art patents were located by our search and copies thereof are enclosed for your consideration:

Patent No.	Issued	Inventor
4,549,291	10/22/85	Renoulin et al.
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5,341,374	08/23/94	Lewen et al.
5,341,418	08/23/94	Yoshida
5,349,640	09/20/94	Dunn et al.
5,392,402	02/21/95	Robrock

> No. 5,392,402 was selected as describing a broadband
> intelligent telecommunications network (BIN) having a fast packet
> (ATM) switch and capable of using the resources of conventional
> circuit-switched telephone networks. As shown in Fig. 4, the
> Line Information Database (LIDB) of the circuit-switched
> telephone network is adapted to be included in the resource
> system 63 for call handling support in the BIN. The resource
> system 63 includes a database 63A in the form of a LIDB 150 which
> is directly connected to a ATM port 70P. The LIDB 150 and the
> LIDB 150B may be connected to circuit-switched network elements
> through SS7 and X.25 links (column 8, line 17, to column 9, line
> 60). The database support of circuit-switched telephone networks
> is thereby made available to fast-packet networks to facilitate
> fast-packet system call handling. "Further, substantial economic
> value can be realized in fast-packet networks by using the
> existing LIDB infrastructure of the circuit-switched telephone
> network" (column 9, lines 15-25).

Yes, this piece of prior art can be an option to implement the intelligence means of the packet-switched network of present invention, if ATM technology is used for that network.

>
> No. 5,058,111 was selected as describing subscriber line
> interface circuits that are between analog telephones and packet
> switched networks, and are capable of reducing the signal-to-
> noise ratio (S/N ratio) both in the analog/digital and the
> digital/analog conversions. Voice signals are transmitted and
> received as analog signals via line terminal circuits which are
> connected to respective packet communication signal processing
> circuits. In the packet communication signal processing
> circuits, packet assembly and disassembly, voice signal
> processing, and line terminal circuit control are carried out in
> synchronism with the packet communication network. Packet data
> are transferred serially between the packet communication signal
> processing circuits and a switch interface circuit 32, which
> interfaces with a packet switched network. Operation of the
> subscriber line interface circuit package 30 is controlled by a
> microprocessor 31.

Yes, the "subscriber line interface circuits" described in No. 5,058,111 may be a technique to implement components of present invention, including voice digitizer, compressor(including silence suppression)/decompressor, packetizer/unpacketizer. However, No. 5,058,111 does not address buffering to reduce packet network-introduced jitters, and optional encryption/decryption issue. Moreover, No. 5,058,111 proposes a one-to-one-coorespondent relationship

etween subscriber lines anu "packet communication sig..al processors, s opposed to shared processors configured on a statistical basis, his can be a waste of resource and thus implies an unnecessarily higher ost.

No. 5,301,189 was selected as describing a telecommunication network having both an asynchronous transfer mode (ATM) exchanges and synchronous transfer mode (STM) exchanges, and wherein interfaces allocated to the ATM exchanges serve the purpose of connecting transmission lines carrying the circuit-switched messages of the STM exchanges. The purpose of the invention is to provide the advantages of telecommunication networks operating in the ATM mode to the older currently installed telecommunication networks operating in the STM mode.

No. 5,301,189 proposes a "shorter cell" format for ATM exchanges. This might be an option to reduce delay time in the packet network part of the present invention. However, using shorter cells raises the overhead and thus reduces the throughput. Also, if anti-jiters buffering introduces a much greater delay, then the savings of shorter cells may not be very beneficial.

>
> No. 5,341,374 was selected as describing a local
> communication network for integrating telephone voice data with
> digital data from other application devices and providing
> distributed call processing for all such data. Where applicable,
> the analog-to-digital conversion and storage operations are
> performed by the telephones 58. Digital data from the telephone
> is sent to an adapter card 70 connected to a node coupling unit
> 51 on a token ring network 50. One or more of the node coupling
> units may be connected to a bridge for providing connection to a
> gateway for an external network, such as a telephone central
> office.

No. 5,341,374 provides a solution for voice communication over pure packet-switched private local area networks, as opposed to packet/circuit hybrid public wide area networks, even though voice packetization techniques are needed for both inventions.

>
> No. 4,549,291 was selected as showing a hybrid local
> communication network that operates both in circuit and packet
> modes. The interface units on the loop 2 are referred to as
> Cluster Control Units and include a microprocessor 417 as shown

in Fig. 4 of this patent. Each cluster control unit may be connected to a plurality of terminals, such as telephones, teleprinters, data terminals, or teletext or videotext receiver sets.

No. 4,549,291 supports multiple types of traffics on one network, as opposed to the present invention which supports one traffic, voice, on two types of networks combined. The same wording "hybrid" appears in both disclosures but means different things.

>
> No. 4,592,048 was selected as disclosing an integrated
> packet switching and circuit switching system comprising a number
> of switching modules each connected to a different plurality of
> user terminals. Each switching module includes a packet
> switching unit used both to provide packet-switched communication
> channels among the user terminals connected to that switching
> module, and to switch control information between the user
> terminals and the control unit to establish both circuit-switched
> calls and packet-switched calls.

No. 4,592,048 supports multiple types of traffics on one switch, as opposed to the present invention which supports one traffic, voice, on two types of networks combined.

>
> No. 4,723,238 was selected as showing an interface circuit
> for interconnecting a packet switch system to a circuit switch
> system, and for enabling a uniform dialing plan to be utilized to
> establish intra-system or inter-system connections. Each
> terminal of the packet switched system is assigned a unique
> terminal address similar to the extension number used for
> terminals of the circuit switched system.

No. 4,723,238 supports data communications with a packet-circuit-packet interworking, as opposed to the present invention, which supports voice communication with a circuit-packet-circuit interworking.

>
> No. 4,926,416 was selected as describing a hybrid packet
> switching network wherein the packets of particular connections,
> such as voice, are treated with priority over packets of other
> connections. In order to avoid delay jitter and loss of
> information, all packets are divided within an exchange into
> subpackets of equal length and distributed to subframes, and
> switching takes place on the basis of the subframes.

o. 4,926,416 proposes a "shorter packets" for ATM/STM exchanges. The
TM packets might be an option to reduce delay time in the packet network
art of the present invention. However, if anti-jitters buffering introduces
much greater delay, then the savings of shorter packets may not be
ery beneficial.

>
> No. 5,014,266 was selected as describing a circuit switching
> system that responds to a switching protocol to set up multiple
> logical links, compressed voice calls and subrate data calls on
> one logical channel between packet switching networks, voice
> concentrators and data multiplexers.

No. 5,014,266 is more related to circuit-switched networking as opposed to packet-switched networking. Although "voice concentrators" mentioned may be a different alternative to better utilize network resources.

>
> No. 5,119,370 was selected as showing a switching node for
> an optical communications network wherein short message traffic
> and connection signaling messages for longer duration and higher
> capacity traffic are both handled by a packet message switch of
> relatively low capacity and rapid response, while the longer
> duration and higher capacity traffic is handled by a circuit
> switch controlled by the signalling messages conducted via the
> packet message switch.

No. 5,119,370 discloses using ATM network to carry circuit-switched network signaling messages and the topic is quite remote to the one in the present invention.

>
> No. 5,251,206 was selected as describing a hybrid packet and
> circuit switching system allowing the merger of packet and
> circuit traffic from a plurality of user interface modules on a
> TDM bus. The data transfers through the hybrid switch are
> performed in bursts for the packet switched traffic as well as
> for the circuit switched traffic. The hybrid switch resolves the
> contentions and grants the transfer on a burst time basis, giving
> to the circuit switched traffic higher priority than for the
> packet switched traffic (column 6, lines 34-50).

No. 5,251,206 discloses a bus structure, which implies a single platform, for carrying both circuit and packet traffics, and no conversion is involved between the two types of traffics. Whereas the present invention

is a distributed system involving conversion between circuit-switched
traffic and packet-switched traffic.

>
>   No. 5,341,418 was selected as describing a terminal adapter
> for providing access from analog signal equipment of the four-
> wire full duplex type to an Integrated Services Digital Network
> System (ISDN).

No. 5,341,418 is about adapters for ISDN, which is still circuit-switched
networking, whereas the present invention is mainly related to carrying
voice through packet-switched networks.

>
>   No. 5,349,640 was selected as describing a ROLM Company
> Digital Telephone having a coder/decoder 130 for converting
> analog voice information to digital form and vice versa, and a
> microprocessor 118 enabling an application operating on this
> computing device to control or monitor operation of the
> telephone.

No. 5,341,418 discloses an adapter which opens up the control
messages of a digital phone internal bus to an external computer and
thus support more powerful control. This is not quite related to the
field of the present invention. However, the digital phone has a function
converting between analog voice and digital voice, and the function
may also be needed by the Gateway Computer in the present invention.

>
>   We would appreciate your reviewing the enclosed patents and
> letting us know the specific features of your disclosure that
> distinguish it from the disclosures of these patents. This will
> help us evaluate the likelihood of successfully prosecuting a
> patent application on your disclosure.
>
>   Please confirm your receipt of this letter by telefax or MCI
> MAIL, and then let us have your comments on the enclosed patents
> as soon as practicable.
>
>   I look forward to hearing from you soon. Please call if you
> have any questions.
>
>   Thank you for your help.
>
>   Cordially,
>

Townsend Belser

TMB/cdc
Enclosures

Advance Copy of Letter Only VIA MCI MAIL

cc via MCI MAIL (Letter Only): Mr. Tim D. Casey
Mr. Alex Huang
Ms. Denise Nappi

EXHIBIT H

Date: [redacted] EST
From: Alex Huang / MCI ID: 727-8033

TO: * Townsend Belser / MCI ID: 729-0130
CC: Tim D. Casey / MCI ID: 201-8157
CC: Denise Nappi / MCI ID: 746-5183
Subject: Re: RIC-95-042; You File: 1643/339
Message-Id: 01950818161410/0007278033PJ3EM

Townsend:

I hope that I can get an e-mail copy of the draft patent application from you.

I was trying to make some comments and suggestions for revisions when I found it would be easier to work on my computer as opposed to hand-writing into the limited space between lines of the original.

Thanks,
Alex

> Date: [redacted] CDT **RECEIPT
> Source-Date: [redacted] EST
> From: Townsend Belser / MCI ID: 729-0130
>
> TO: * Alex Huang / MCI ID: 727-8033
> CC: Tim D. Casey / MCI ID: 201-8157
> CC: Denise Nappi / MCI ID: 746-5183
> Subject: RIC-95-042; Our File: 1643/339
> Message-Id: 50950731205005/0007290130PJ4EM
>
>
> [redacted]
>
>
> VIA FEDERAL EXPRESS
> Mr. Lisheng "Alex" Huang
> MCI Communications Corporation
> 2400 N. Glenville Drive
> Richardson, Texas 75082
>
> Re: U.S. Patent Application for
> HYBRID PACKET-SWITCHED AND
> CIRCUIT-SWITCHED TELEPHONY
> Your Ref: RIC-95-042
> Our Ref: 1643/339
>

>
> Enclosed for your review and approval is a draft patent
> application (23 pages of specification and 4 informal sheets of
> drawings) on your above invention disclosure.
>
> Please go over the enclosed application and drawings
> carefully and let us know by telefax or MCI Mail of any changes
> needed to put it in shape for filing in the U.S. Patent and
> Trademark Office. Please do this within the next few days as
> MCI's management has asked us to file this application as soon as
> possible.
>
> If any revisions are needed, your comments and suggestions
> can be written on the enclosed copy which then should be returned
> to us by telefax. In this regard, please insert any additional
> disclosure that may be useful in further distinguishing the prior
> art patents enclosed with my letter to you of [redacted]
>
> The specification, which concludes with a series of numbered
> claims, should accurately describe and claim the invention. If,
> after reading the application, you have any questions about the
> accuracy of the specification or drawings, or whether the claims
> adequately define the invention, please raise those questions
> with me by calling me at (202) 331-7111.
>
> The law requires that a patent application contain a
> detailed description of the best mode contemplated by the
> inventor(s) for carrying out the invention, and that the
> description be complete enough for a person skilled in this art
> to make and use the invention. Please let us know if you feel
> that there might be anything missing in this regard.
>
> Your careful attention to each of the above matters will
> help us ensure that the patent application filed is in the best
> possible condition for examination by the Patent Office and for
> issuance as a patent if the application is allowed.
>
> Please call me at the above telephone number if you have any
> questions regarding any of the above matters.
>
> We also need to know your residence address and citizenship.
>
> Thanking you for your assistance in this matter,
>
> Cordially,
>
> POLLOCK, VANDE SANDE & PRIDDY
>
>

>
> Townsend M. Belser, Jr.
>
> TMB/cdc
> Enclosure: Draft patent application
>
> Copy of letter VIA MCI MAIL - Tim D. Casey
> Alex Huang
> Denise Nappi
>

EXHIBIT I

file

Date: [redacted] EDT
From: Tim D. Casey / MCI ID: 201-8157

TO: * Jodette Kaspar / MCI ID: 203-6509
Subject: RIC-95-042, 1643/339
Message-Id: 75950824135757/0002018157PL4EM

Forwarded message:

Date: [redacted] EDT **RECEIPT
Source-Date: [redacted] EST
From: Townsend Belser / MCI ID: 729-0130

TO: Alex Huang / MCI ID: 727-8033
CC: * Tim D. Casey / MCI ID: 201-8157
Subject: RIC-95-042, 1643/339
Message-Id: 33950821164733/0007290130PJ1EM

Alex, per your e-mail request of [redacted], set forth below is an e-mail copy of the draft patent application on the subject disclosure. I look forward to receiving your comments and suggestions for revisions as soon as practicable.

Thank you for your help, Townsend

HYBRID PACKET-SWITCHED AND CIRCUIT-SWITCHED TELEPHONY SYSTEM

TECHNICAL FIELD

This invention relates to telecommunication systems, and, more particularly, to a hybrid telephony system comprising both circuit-switched and packet-switched networks.

BACKGROUND OF THE INVENTION

With the extensive use of personal computers and other data processing facilities both at home and in the office, a need exists for providing integrated voice and data transmission and switching capabilities on a widespread basis. This has led to the development of an integrated service digital network (ISDN), which is a switched communications network providing end-to-end digital connectivity among network users where voice and data services are provided over the same transmission and switching facilities.

Voice and data traffic have significantly different characteristics. Voice is typically continuous in one direction for relatively long intervals and tolerant of noise, but

sensitive to variations in delay. Data is bursty and sensitive to noise errors, but tolerant of moderate delays and variations in arrival times.

Two fundamental different switching techniques have therefore been traditionally applied to voice and data transmissions. Circuit switching, where switched connections between users are dedicated for call duration, is the basis of the present-day switched voice telecommunication network. On the other hand, packet switching, where data packets from multiple terminals share a single, high-speed line and are switched based on logical channel numbers attached in the packets, is being rapidly adopted as the basis of the present-day switched data telecommunication network.

Packet switching was pioneered in the ARPANET network of the U.S. Department of Defense, and has been widely implemented in a variety of public data networks. However, most public telephone systems are fundamentally circuit switched, which is an inherently inefficient system because typically each subscriber utilizes the allotted channel for a relatively small amount of the total time. Furthermore, the number of simultaneous circuit-switched communications are limited because only a portion of the available bandwidth is allocated to such communications.

Another disadvantage is that, because circuit switching is centralized, a failure at the switching center can result in failure of the entire network. A further disadvantage of circuit-switched telephony is due to the proprietary nature of the telephony switches currently in use. Because the switching software is often proprietary and not shared with other manufacturers, the cost and delay in adding and interfacing new services are often frustrating and installation prohibiting.

It has been proposed that packet-switched techniques be combined with at least some circuit-switched telephony so that the entire system bandwidth may be made available to each subscriber on a random access basis. For this purpose, there are currently emerging software products that make use of the Internet, which is a constantly changing collection of interconnected packet-switched networks, for telephony. VOCALTEC software provides half-duplexed long-distance telephone capability through the Internet. Camelot Corp is another entry in the Internet telephone business with a MOSAIC front end software that supports full-duplexed voice conversation. These products offer an alternative to long-distance analog telephone service for the subscribers by digitizing and compressing voice signals for transport over the Internet.

Some limitations of this type of hybrid telephone system are: (1) Both the caller and the callee must have computers compatible with the Internet, (2) they must have sound systems on

their computers, (3) they must have full Internet access, and (4) they must be both running compatible software at the time the call is made. These limitations translate into a considerable amount of investment in hardware and software, which has to be made by the individual subscribers to implement such a telephony system. The last limitation also means that the calls have to be scheduled in advance in most cases, which clearly does not provide the convenience of conventional telephone calls. An additional problem with such software products is that the performance is constrained by the CPU capabilities of each computer, such as processor speed, memory capacity, and I/O processing.

SUMMARY OF THE INVENTION

In accordance with the principles of the present invention, a hybrid packet-switched and circuit-switched telephony (HPCT) system routes a telephone call mostly through packet-switched networks, except for the caller and callee ends where the subscriber telephone sets are directly connected to the circuit-switched networks of the respective local exchange companies (LEC's). A Gateway Computer (GC) or equivalent interconnects the packet-switched network to each of the circuit-switched networks, and converts voice signals into data packets and vice versa, and resolves the call destinations while routing the packets.

In this invention, the GCþs are preferably managed by the telephony service provider, as opposed to the end-user. Because the GCþs are a set of resources shared by many subscribers, they can be managed with higher efficiency and utilization than calls managed by a subscriberþs personal computer. By incorporating the HPCT system into the current long-distance telephony, lower cost of communication can be achieved due to better utilization of available channels by packet-switched networks over purely circuit-switched networks, and the benefits of packet switching can be made available to many subscribers without significant subscriber investment.

Additional advantages of this hybrid packet and circuit switched telephony are: (1) lower cost of transport due to better circuit utilization as compared to a pure circuit-switched network; (2) availability to any subscriber at no initial investment as would be required by pure packet-switched telephony, such as requiring purchase of a multimedia personal computer and Internet access; (3) the potential for quickly adding intelligent services due to computer based telephony, such as a caller's personalized speed dialing list, a callee's personalized virtual destination number, and integration with electronic mails; (4) substantial reductions in access and transmission costs as compared to pure circuit-switched

telephony; and (5) avoidance of the inconvenience of current packet-switched telephony with Internet phones.

BRIEF DESCRIPTION OF THE DRAWINGS

The features of the invention and its objects and advantages may be further understood from the detailed description below taken in conjunction with the accompanying drawings, in which:

Fig. 1 is a block diagram showing a first embodiment of the present invention;

Fig. 2 is a block diagram illustrating a voice telephony system before and after incorporating the first embodiment of the present invention;

Fig. 3 is a system block diagram of the first embodiment illustrating operation of the Gateway Computer;

Fig. 4 is a block diagram showing a second embodiment of the present invention;

Fig. 5 is a flow diagram of the calling process for providing a charge call in accordance with the present invention;

Fig. 6 is a sequence diagram of the call signalling for providing the charge call of Fig. 5; and,

Fig. 7 is a flow diagram of the calling process for providing a call from a caller's dedicated telephone in accordance with the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

With reference to Figs. 1 to 3, a hybrid packet-switched and circuit-switched telephony (HPCT) system according to a preferred embodiment of the present invention comprises originating and terminating gateway computers (GCþs) which interconnect corresponding circuit-switched networks with a packet-switched network for voice and data communications.

As shown in Fig. 1, an originating (local) telephone set 1 is connected with an originating (local) GC 3 through a circuit-switched network 2 of an originating local exchange carrier (LEC). At the other end of the telephony system, a terminating (remote) telephone set 8 is connected with a terminating (remote) GC 6 through a terminating (remote) circuit-switched network 5 of a terminating (remote) LEC 7. A packet-switched network 5 is provided for communications between originating GC 3 and terminating GC 6. Fig. 2 shows diagrammatically how a conventional circuit-switched network 10 is replaced by the two GC's 3 and 6 and the packet-switched network 5.

Although only shown for one of the GC's in Fig. 3, preferably both the originating and terminating GCþs further include a voice prompt playback unit 3.1, an address resolution unit 3.2, a data manipulator 3.3, a dialed digit receiver 3.4 and a spoken digit recognizer 3.5. The data manipulator 3.3 has many

important functions in the hybrid telephony system where it converts data between various formats to accommodate both circuit and packet switched networks. The data manipulator 3.3 thus comprises a voice stream digitizer, a packetizer/unpacketizer and a stream buffer, and optionally may further comprise a compressor/decompressor and a encryptor/decryptor.

With a GC having these functions, the expected voice compression ratio may reach 25:1, or even better with emerging technology. The presence of the usual amounts of silence in voice may double that ratio to 50:1, making the HPCT even more efficient and cost-effective. To achieve even further compression ratios, special compression schemes may also be used, which are expected to be both tolerated by the human ear and used to facilitate a low cost of the service. The HPCT may provide a virtual end-to-end connection. In the absence of such a virtual connection, the buffering mechanism at the receiving end can recover the stream of voice from packets arriving with the variable delay introduced by the packet-switched network.

The HPCT telephony network of Fig. 1, along with its associated service protocols, is symmetric. However, sometimes a caller may have a multimedia-capable computer and a packet-switched network connection, thereby enabling advanced services or features. However, if the callee does not have (a) a terminating multimedia computer, (b) direct access to the packet-switched network, and (c) a compatible packet-switched telephony application currently running on the terminating computer, then the call can not be completed over just a packet-switched network. In this case, a terminating circuit-switched LEC 7 supported by a terminating GC 6 may be used in the same way as in the first embodiment of the present invention, but the telephony system will have an asymmetric configuration as shown in Fig.4.

In this system, the callerþs multimedia computer 4 will run a digital communications program comparable to an originating GCþs protocol and therefore will serve as the originating GC from the view point of the packet-switched network 5 and the terminating GC 6, except the billing and validation of the caller may be performed by the terminating GC 6 based on the callerþs access point to the packet-switched network. Similarly, where the callee has a multimedia capable computer and a packet-switched network connection but the caller does not, the telephony system of the invention may have an asymmetric configuration that is the reverse of the Fig. 4 configuration.

On the other hand, where the HPCT utilizes Gateway Computers at both ends of the packet-switched network, each GC provides a set of resources that are shared by many users and thus achieves much higher utilization in the telephony than a personal computer. Optimization of performance can be achieved by using

Digital Signal Processors (DSP's) or Application Specific Integrated Circuits (ASIC's). Furthermore, the users do not have to make a large investment, operate special computer equipment and programs, or schedule calls in advance. In fact, as described relative to Figs. 5 and 7, the users may not tell any difference between using the HPCT and using their regular long-distance services, except for a much lower cost.

Considering the 50:1 compression ratio discussed above, the utilization of the circuits in circuit-switched telephony can be only 1/50 as efficient as that of the HPCT; in other words, the cost of the former can be as much as 50 times higher than that of the HPCT. Another important problem with circuit-switched telephony is the proprietary nature of the telephony switches which are the foundation of this telephony. Because switch software development is only done by the manufacturers, the cost and delay in adding new services are often frustrating and prohibiting. The HPCT, however, is based on general-purpose computers with open architecture, which can open up development and bring very cost-effective new services in a much shorter time frame.

The packet-switched network 5 of the HPCT system can be one of many types of packet-switched public data networks, such as X.25 or the emerging Asynchronous Transfer Mode (ATM) network. The ATM network is a special packet-switched network with low delay and low delay deviation, in which data is formatted into special types of packets, referred to as þcellsþ, to achieve fast-switching. Accordingly, ATM networks are sometimes referred to as having a third type of networking, namely "cell-switched networking".

A caller can use the HPCT system as an alternative long distance telephony service (þcharge serviceþ), or the caller can use it as his/her primary long distance telephony service (þdedicated serviceþ). Charge service can be reached from any telephone while dedicated service can be reached only from a subscriberþs dedicated telephone, such as a home phone or office phone. The alternative service is referred as þcharge serviceþ because the caller does not need to have a dedicated telephone account with the service provider; instead, the authorization is via a credit card or calling card inquiry.

To implement the charge service and the dedicated service, the invention provides two respective protocols for processing calls within the hybrid telephony system. A first protocol for the charge service is illustrated in Figs. 5 and 6. This process includes the following steps: (1) The caller first calls a local originating GC through a circuit-switched originating LEC from any telephone, and the callerþs address (callerþs telephone number) is relayed to the originating GC by the originating LEC.

(2) The originating GC plays a voice prompt (a greeting message asking for input) to the caller asking for the calleeþs destination address (calleeþs telephone number). (3) The caller provides the address either through telephone keypad digits or through spoken digits which are recognized by the originating GC. (4) The originating GC resolves the call routing information in a manner similar to the Domain Name Service for the Internet, obtains the packet network address (such as the IP address of the Internet) of the terminating GC, which is usually local to the callee (otherwise a toll call may be involved), and estimates the unit charge for a call going through that terminating GC. (5) The originating GC informs the caller about the charge rate, and asks for the callerþs preferred payment method, such as by credit card, or through a prearranged calling card account. (6) The caller specifies the payment method either through keypad digits or through spoken digits which again are recognized by the originating GC (if this is a collect call, then the callerþs spoken information about both parties is recorded and digitized to be announced later to the callee). (7) The originating GC validates the payment method through an internal or external database. (8) The originating GC sends a control message to the terminating GC, along with both partyþs addresses and, if the terminating GC does not know where to route the call or does not have the resource to serve the call, it responds with a negative acknowledgment and an alternative terminating GC is searched for and selected, or the caller is informed that the call can not be routed at that moment. (9) The terminating GC dials out to the callee through a circuit switched terminating LEC using the destination address it obtained from the originating GC. (10) If the call proceeds successfully through the terminating LEC, the terminating GC sends an acknowledgment back to the originating GC (or if the call proceeds unsuccessfully, such as due to busy telephone lines, the terminating GC sends this status to the originating GC in the form of a busy message). (11) The originating GC then passes the status of acknowledgment back to the caller through the originating LEC, the effect being a ring back tone (or a busy tone). (12) The callee answers the call. (13) The terminating GC passes this state change to the originating GC, which may begin billing at that time. (14) The callee starts the conversation by greeting the caller. (15) The terminating GC continuously digitizes all the voice signals from the callee, and, after possibly encrypting and compressing, packetizes the data into packet form, the packets then being sent over the packet-switched network to the originating GC. (16) The originating GC, after possibly rearranging the packets to maintain proper packet order, unpacketizes the received data and, after possibly decompressing and decrypting, converts the

digitized data back to the voice signal, which is then routed to the caller over the circuit-switched network of the originating LEC. (17) The same process as described in steps 14 through 16 is performed for the callerþs voice in the opposite direction. The resulting processing in both directions supports the conversation between the two parties participating in the call.

The GC's preferably provide out-of-band signaling, and the call signaling sequence for providing the charge call of Fig. 5 will now be described with reference to Fig. 6. (1) The caller's telephone number is sent to the originating GC to access a call. (2) The originating GC prompts for a destination address, such as by a dial tone. (3) The caller inputs the callee's address, such as by dialing the callee's telephone number. (4) The originating GC may provide a voice message regarding rate, and prompts for a payment method, such as by a special tone or by a voice message. (5) The caller inputs the desired method of payment, such as keypad numbers corresponding to a credit card. (6) The originating GC validates the payment method and then sends a connection request to the terminating GC. (7) The terminating GC dials the callee through the terminating LEC. (8) Depending on whether the callee's telephone is available or busy, the terminating GC sends a corresponding acknowledgement to the originating GC. (9) Depending on whether the callee's telephone is available or busy, the originating GC sends a ring back tone or a busy signal to the caller's telephone. (10) If the callee answers the telephone, an off-hook signal is sent to the terminating GC. (11) The terminating GC then sends an answer indication to the originating GC, which starts billing and sets up the in-band routing for both digitized voice data and analog voice transmission. (12) At this stage, either the callee or the caller may initiate the conversation. If initiated by the callee, the callee's telephone sends the voice greeting to the terminating GC. (13) The terminating GC digitizes the analog voice, and may also perform the additional functions previously described, and then sends the digitized voice data to the originating GC. (14) The originating GC converts the voice data to analog voice by performing the functions previously described, and sends the analog voice to the caller.

A second protocol for the dedicated service on the HPCT telephony system of the invention is shown in Fig.7. This process includes the following steps: (1) The caller initiates a long-distance call by dialing a destination address (calleeþs telephone number) through the circuit-switched network of an LEC from his/her dedicated telephone, such as a home phone or office phone, for which a routing configuration to an originating GC is present within the LEC. The LEC routes the call to the GC, and the callerþs address (callerþs telephone number) is passed to the

GC by the LEC, along with the destination address. (2) The originating GC authorizes the call by checking the callerþs account information through an internal database or may communicate with a centralized database for the account information, and it also resolves the call routing information using the dialed destination address. (3) The originating GC then sends a control message to a terminating GC, along with both partyþs addresses. If the first terminating GC does not know where to route the call or does not have the resources to service the call, it responds with a negative acknowledgment and an alternate terminating GC is searched for and selected, or the caller is informed of the negative acknowledgement. (4) The terminating GC dials out to the callee through the circuit-switched network of the terminating LEC. (5) If the call proceeds successfully through the terminating LEC, the terminating GC sends an acknowledgment back to the originating GC (the handling of the unsuccessful, most likely busy, scenario is similar to that in the first protocol). (6) The originating GC then passes the status back to the caller through the originating LEC, the effect being a ring-back tone. (7) The callee answers the call. (8) The terminating GC passes this state change to the originating GC which may start charging. (9) The callee initiates the conversation by greeting the caller. (10) The terminating GC continuously digitizes all the voice signals from the callee, possibly encrypts and compresses, and packetizes the data into packets, the packets then being sent over the packet-switched network to the originating GC. (11) The originating GC, after possibly rearranging the packets to maintain proper packet order, unpacketizes the data, possibly decompresses and decrypts, and converts it back to the voice signal, which is then routed to the caller over the circuit-switched network of the originating LEC. (12) The same process described in steps 9 through 11 is performed for the callerþs voice, except in the opposite direction. This processing in both directions supports the conversation between the two parties participating in the call.

While the present invention has been described in connection with a system having a circuit-switched network in the form of an analog local exchange carrier serving analog telephone sets, it will be appreciated that many of the features of the present invention are also applicable for use with digital telephones and a public circuit-switched data network. With a digital circuit-switched network, the configuration of the corresponding GCþs could be simplified since it is no longer necessary for the data manipulator to convert the voice signal into digital data, and vice versa.

From the foregoing, it can be seen that the present invention provides an improved telephony system which effectively

integrates voice and data in a hybrid circuit-switched and packet-switched telephony network, while ensuring real-time high quality voice communication and calling services with low transmission and access cost. By utilizing gateway computers of telephony service providers to route calls between circuit-switched and packet-switched telephone networks, HPCT provides the benefits of packet switching to any telephone subscriber, with none of the substantial initial investments required by pure packet-switched telephony. The potential for vastly increased intelligent services due to computer based telephony, such as callerþs personalized speed dialing list, calleeþs personalized virtual destination number, integration with electronic mails, and many others, allows for even further enhancement of the HPCT system. Furthermore, the HPCT system, which is based on general-purpose computers with open architecture, can open development of a host of new services and make them cost-effective in a much shorter time than would be required for complete conversion from conventional circuit-switched networks to entirely packet-switched networks.

While this invention has been described in the context of preferred embodiments comprising at least one circuit-switched network of an LEC, it should be clear that the principles of the invention will work equally well with other telecommunications networks and with variations of the preferred embodiment that will be apparent to those skilled in the art. Many other modifications and alternatives are possible and will occur to those skilled in the art who become familiar with the instant disclosure. Such modifications and alternatives are intended to be within the scope of the invention as defined by the claims set forth below.

CLAIMS

What is claimed is:

1. A telecommunications system comprising:

an originating circuit-switched network for transmitting originating voice signals;

originating gateway means for converting said originating voice signals into packets of digital data,

terminating gateway means for converting said digital packets into terminating voice signals,

a terminating circuit-switched network for transmitting said terminating voice signals, and

a packet-switched network for transmitting said digital packets from said originating gateway means to said terminating gateway means, at least one of said originating and terminating gateway means comprising means for routing said digital packets through said packet-switched network from said originating gateway means to said terminating gateway means.

2. A telecommunications system according to claim 1, wherein said originating gateway means comprises means for compressing said digital data, and wherein said terminating gateway means comprises means for decompressing said digital data.

3. A telecommunications system according to claim 1, wherein said originating gateway means comprises means for encrypting said digital data, and wherein said terminating gateway means comprises means for decrypting said digital data.

4. A telecommunications system according to claim 1, wherein said terminating gateway means comprises terminating buffer means for storing said digital packets prior to the conversion thereof into said terminating voice signals.

5. A telecommunications system according to claim 4, wherein said terminating gateway means further comprises means for rearranging said stored digital packets to maintain a proper packet order.

6. A telecommunications system according to claim 1, wherein said routing means provides said routing in response to dialled digits.

7. A telecommunications system according to claim 1, wherein said routing means provides said routing in response to spoken digits.

8. A telecommunications system according to claim 1, wherein said terminating circuit-switched network comprises means for transmitting first return voice signals to said terminating gateway means; wherein said terminating gateway means comprises means for converting said first return voice signals into packets of return digital data; wherein at least one of said originating and terminating gateway means comprises means for routing said

return packets through said packet-switched network from said terminating gateway means to said originating gateway means; and wherein said originating gateway means comprises means for converting said return packets into second return voice signals.

9. A telecommunications system according to claim 8, wherein said originating gateway means comprises originating buffer means for storing said return packets prior to conversion thereof into said second return voice signals.

10. A telecommunications system according to claim 9, wherein said originating gateway means further comprises means for rearranging said stored return packets to maintain a proper packet order.

11. A telecommunications system comprising:

originating means for providing digital packets corresponding to originating voice signals,

terminating gateway means for converting said digital packets into terminating voice signals,

a terminating circuit-switched network for transmitting said terminating voice signals, and

a packet-switched network for transmitting said digital packets from said originating means to said terminating gateway means, at least one of said originating means and said terminating gateway means comprising means for routing said digital packets through said packet-switched network from said originating means to said terminating gateway means.

12. A telecommunications system according to claim 11, wherein said originating means comprises means for compressing said digital data, and wherein said terminating gateway means comprises means for decompressing said digital data.

13. A telecommunications system according to claim 11, wherein said originating means comprises means for encrypting said digital data, and wherein said terminating gateway means comprises means for decrypting said digital data.

14. A telecommunications system according to claim 11, wherein said terminating gateway means comprises terminating buffer means for storing said digital packets prior to the conversion thereof into said terminating voice signals.

15. A telecommunications system according to claim 14, wherein said terminating gateway means further comprises means for rearranging said stored digital packets to maintain a proper packet order.

16. A telecommunications system according to claim 11, wherein said routing means provides said routing in response to data received from said terminating gateway means.

17. A telecommunications system according to claim 11, wherein said routing means provides said routing in response to a typed input from a computer keyboard.

18. A telecommunications system according to claim 11, wherein said terminating circuit-switched network comprises means for transmitting first return voice signals to said terminating gateway means; wherein said terminating gateway means comprises means for converting said first return voice signals into packets of return digital data; wherein at least one of said originating means and said terminating gateway means comprises means for routing said return packets through said packet-switched network from said terminating gateway means to said originating means; and wherein said originating means comprises means for converting said return packets into second return voice signals.

19. A telecommunications system according to claim 18, wherein said originating means comprises originating buffer means for storing said return packets prior to conversion thereof into said second return voice signals.

20. A telecommunications system according to claim 19, wherein said originating means further comprises means for rearranging said stored return packets to maintain a proper packet order.

21. A telecommunications method comprising steps of:

providing digital packets for transmission from an originating means, said digital packets corresponding to originating voice signals,

transmitting said digital packets from said originating means to a terminating gateway means through a packet-switched network, at least one of said originating means and said terminating gateway means comprising means for routing said digital packets through said packet-switched network from said originating means to said terminating gateway means,

converting said digital packets into terminating voice signals for transmission from said terminating gateway means, and

transmitting said terminating voice signals through a terminating circuit-switched network.

ABSTRACT OF THE DISCLOSURE

A telecommunication system includes an originating circuit-switched network for transmitting originating voice signals, an originating gateway computer for converting the originating voice signals into packets of digital data, a terminating gateway computer for converting the digital packets into terminating voice signals, a terminating circuit-switched network for transmitting the terminating voice signals, and a packet-switched network for transmitting the digital packets from the originating gateway computer to the terminating gateway computer. Either the originating or terminating gateway computer may route the digital packets through the packet-switched network from one computer to the other. At either the originating or terminating end of the system, the corresponding circuit-switched network and gateway computer may be replaced by a computer terminal running a digital communications program.

EXHIBIT J

ate: [redacted]
rom: Alex Huang / MCI ID: 727-8033

O: * Townsend Belser / MCI ID: 729-0130
C: Tim D. Casey / MCI ID: 201-8157
C: Denise Nappi / MCI ID: 746-5183
ubject: Re: RIC-95-042; Your File: 1643/339
essage-Id: 60950830174106/0007278033PJ4EM

'ownsend,

nclosed please find my recommended revision of the patent application
hich you drafted and sent to me previously.

t the same time, I am separately faxing to you all the drawings, including two
ew diagrams. By the way, if your office happens to use Micrografx
esigner drawing software, I can e-mail you my drawings electronically.

'or marking where I have additions or modifications, I am also faxing
o you a working version of the revision, in it the deletions
re shown as strikethrough text, the addition underlined, and changed
paragraphs with side bars.

Thank you very much for the impressive draft and all other help.

Alex

HYBRID PACKET-SWITCHED AND CIRCUIT-SWITCHED TELEPHONY SYSTEM

TECHNICAL FIELD

This invention relates to telecommunication systems, and, more particularly, to a hybrid telephony system comprising both circuit-switched and packet-switched networks.

OTHER PUBLICATIONS

P. Vary et al. "Speech codec for the European mobile radio system", IEEE GLOBECOM 1989, Nov. 1989.

BACKGROUND OF THE INVENTION

With the extensive use of personal computers and other data processing facilities both at home and in the office, and hence great needs for data communications, packet-switched public networks become rapidly developed and increasingly interconnected with each other.

RIC-95-042
file

Date: [redacted] EDT
From: Tim D. Casey / MCI ID: 201-8157

TO: * Jodette Kaspar / MCI ID: 203-6509
Subject: Re: RIC-95-042; Your File: 1643/339
Message-Id: 24950905140442/0002018157PL2EM

Forwarded message:

Date: [redacted] EDT
From: Alex Huang / MCI ID: 727-8033

TO: Townsend Belser / MCI ID: 729-0130
CC: * Tim D. Casey / MCI ID: 201-8157
CC: Denise Nappi / MCI ID: 746-5183
Subject: Re: RIC-95-042; Your File: 1643/339

Townsend,

Enclosed please find my recommended revision of the patent application which you drafted and sent to me previously.

At the same time, I am separately faxing to you all the drawings, including two new diagrams. By the way, if your office happens to use Micrografx Designer drawing software, I can e-mail you my drawings electronically.

For marking where I have additions or modifications, I am also faxing to you a working version of the revision, in it the deletions are shown as strikethrough text, the addition underlined, and changed paragraphs with side bars.

Thank you very much for the impressive draft and all other help.

Alex

HYBRID PACKET-SWITCHED AND CIRCUIT-SWITCHED TELEPHONY SYSTEM

TECHNICAL FIELD

This invention relates to telecommunication systems, and, more particularly, to a hybrid telephony system comprising both circuit-switched and packet-switched networks.

OTHER PUBLICATIONS

P. Vary et al. "Speech codec for the European mobile radio system", IEEE GLOBECOM 1989, Nov. 1989.

BACKGROUND OF THE INVENTION

With the extensive use of personal computers and other data processing facilities both at home and in the office, and hence great needs for data communications, packet-switched public networks become rapidly developed and increasingly interconnected with each other. These packet networks have mostly been serving data communications traffics as opposed to voice telephony.

Voice and data traffic have significantly different characteristics. Voice is typically continuous in one direction for relatively long intervals and tolerant of noise, but sensitive to variations in delay. Data is bursty and sensitive to noise errors, but tolerant of moderate delays and variations in arrival times.

Two fundamental different switching techniques have therefore been traditionally applied to voice and data transmissions. Circuit switching, where switched connections between users are dedicated for call duration, is the basis of the present-day switched voice telecommunication network. On the other hand, packet switching, where data packets from multiple terminals share a single, high-speed line and are switched based on logical channel numbers attached in the packets, is being rapidly adopted as the basis of the present-day switched data telecommunication network.

Packet switching was pioneered in the ARPANET network of the U.S. Department of Defense, and has been widely implemented in a variety of public data networks. However, most public telephone systems are fundamentally circuit switched, which is an inherently inefficient system because typically each subscriber utilizes the allotted channel for a relatively small amount of the total time. Furthermore, the number of simultaneous circuit-switched communications are limited because only a portion of the available bandwidth is allocated to such communications.

Another disadvantage is that, because circuit switching is centralized, a failure at the switching center can result in failure of the entire network. A further disadvantage of circuit-switched telephony is due to the proprietary nature of the telephony switches currently in use. Because the switching software is often proprietary and not shared with other manufacturers, the cost and delay in adding and interfacing new services are often frustrating and installation prohibiting.

It has been proposed that packet-switched techniques replace, or at least be combined with some, circuit-switched telephony so that the entire system bandwidth may be made available to each subscriber on a random access basis. For this purpose, there are

currently emerging software products that make use of the Internet, which is a constantly changing collection of interconnected packet-switched networks, for telephony. VOCALTEC software provides half-duplexed long-distance telephone capability through the Internet. Camelot Corp is another entry in the Internet telephone business with a MOSAIC front end software that supports full-duplexed voice conversation. These products offer an alternative to long-distance analog telephone service for the subscribers by digitizing and compressing voice signals for transport over the Internet.

Some limitations of this type of hybrid telephone system are: (1) Both the caller and the callee must have computers, (2) they must have sound systems on their computers, (3) they must have full Internet access, (4) they must have both purchased compatible software (5) they must both connect to the Internet at the time the call is made, and 6) the telephony software must be in execution at both ends at the same moment. These limitations translate into a considerable amount of investment in hardware and software, which has to be made by the individual subscribers to implement such a telephony system. The last limitation also means that the calls have to be scheduled in advance in most cases, which clearly does not provide the convenience of conventional telephone calls. An additional problem with such software products is that the performance is constrained by the capabilities of each computer, such as processor speed, memory capacity, and modem functional features.

SUMMARY OF THE INVENTION

In accordance with the principles of the present invention, a hybrid packet-switched and circuit-switched telephony (HPCT) system routes a telephone call mostly through packet-switched networks, except for the caller and callee ends where the subscriber telephone sets are directly connected to the circuit-switched networks of the respective local exchange companies (LEC's). A Gateway Computer (GC) or equivalent interconnects the packet-switched network to each of the circuit-switched networks, and converts voice signals into data packets and vice versa, and resolves the call destinations while routing the packets.

In this invention, the GC's are preferably managed by the telephony service provider, as opposed to the end-user. Because the GC's are a set of resources shared by many subscribers, they can be managed with higher efficiency and utilization than calls managed by a subscriber's personal computer. By incorporating the HPCT system into the current long-distance telephony, lower cost of communication can be achieved due to better utilization of available channels by packet-switched networks over purely circuit-switched networks, and the benefits of packet switching

can be made available to many subscribers without significant subscriber investment. Moreover, there can be special hardware components added to the GC to improve performance, such as DSP or ASIC based compressor, decompressor, speech recognizer, encryptor and decryptor, etc., which would be less cost-effective to added to home computers.

Additional advantages of this hybrid packet and circuit switched telephony are: (1) lower cost of transport due to better circuit utilization as compared to a pure circuit-switched network; (2) availability to any subscriber at no initial investment as would be required by pure packet-switched telephony, such as requiring purchase of a multimedia personal computer, Internet access, and Internet telephony software; (3) the potential for quickly adding intelligent services due to computer based telephony, such as a caller's personalized speed dialing list, a callee's personalized virtual destination number, and integration with electronic mails; (4) substantial reductions in transmission costs as compared to pure circuit-switched telephony; and (5) avoidance of the inconvenience of current packet-switched telephony with Internet phones, including the burden of learning callee's IP address, carrying a portable computer when travel or commute.

BRIEF DESCRIPTION OF THE DRAWINGS

The features of the invention and its objects and advantages may be further understood from the detailed description below taken in conjunction with the accompanying drawings, in which:

Fig. 1 is a block diagram showing a first embodiment of the present invention;

Fig. 2 is a block diagram illustrating a voice telephony system before and after incorporating the first embodiment of the present invention;

Fig. 3a is a system block diagram of the first embodiment illustrating operation of the Gateway Computer which is presented as a structural decomposition;

Fig. 3b is a system block diagram of the first embodiment illustrating operation of the Gateway Computer which is presented as a functional decomposition;

Fig. 4 is a block diagram showing a second embodiment of the present invention;

Fig. 5 is a flow diagram of the calling process for providing a charge call in accordance with the present invention;

Fig. 6 is a sequence diagram of the call signaling for providing the charge call of Fig. 5; and,

Fig. 7 is a flow diagram of the calling process for

providing a call from a caller's dedicated telephone in accordance with the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

With reference to Figs. 1 to 3, a hybrid packet-switched and circuit-switched telephony (HPCT) system according to a preferred embodiment of the present invention comprises originating and terminating gateway computers (GC's) which interconnect corresponding circuit-switched networks with a packet-switched network for voice and data communications.

As shown in Fig. 1, an originating (local) telephone set 1 is connected with an originating (local) GC 3 through a circuit-switched network 2 of an originating local exchange carrier (LEC). At the other end of the telephony system, a terminating (remote) telephone set 8 is connected with a terminating (remote) GC 6 through a terminating (remote) circuit-switched network 5 of a terminating (remote) LEC 7. A packet-switched network 5 is provided for communications between originating GC 3 and terminating GC 6. Fig. 2 shows diagrammatically how a conventional circuit-switched network 10 is replaced by the two GC's 3 and 6 and the packet-switched network 5.

Although only shown for one of the GC's in Fig. 3a, preferably both the originating and terminating GC's further include a plurality of optional analog-to-digital and digital-to-analog converter pairs 5, a plurality of digital trunk interfaces 6, a Direct Random Access Memory (DRAM) 7, a signaling network interface 8, a non-blocking Time-Division Multiplexing (TDM) bus 9, a plurality of packet network interface 10, a single, or plurality of, Central Processing Unit (CPU) 11, a plurality of Digital Signal Processor (DSP) 12, a plurality of Application-Specific Integrated Circuit (ASIC) 13, and a plurality of disk controller 14 with disks 15, and a system back plane in the form of either a shared bus or cross connection 19.

Fig. 3b shows the functional components of GC, including a spoken digit recognizer 14 implemented with DSP, a voice prompt playback unit 17 implemented with DSP, an address resolution logic 15 implemented with CPU, a network routing database implemented with CPU and possiblly shared with other GCs in a distributed manner, a tone detector 11 implemented with DSP or ASIC for both user keypad dialing and in-band signaling if needed, a tone generator implemented with DSP or ASIC for prompting and in-band signaling if needed, an optional plurality of analog-to-digital and digital-to-analog converter pairs 10, a plurality of channelized voice bit stream buffer 2 implemented with DSP or ASIC, a compressor/decompressor 3 implemented with DSP or ASIC, a plurality of hardware supervision logic 4 implemented with digital trunk interfaces, a packetizer/unpacketizer 6 implemented with CPU, DSP or ASIC, and an optional encryptor/decryptor 7 implemented with CPU, DSP or ASIC.

With a GC having these functions, the expected voice

compression ratio may reach 25:1, or even better with emerging technology. The presence of the usual amounts of silence in voice may double that ratio to 50:1, making the HPCT even more efficient and cost-effective. To achieve even further compression ratios, special compression schemes may also be used, which are expected to be both tolerated by the human ear and used to facilitate a low cost of the service. The HPCT may provide a virtual end-to-end connection. In the absence of such a virtual connection, the buffering mechanism at the receiving end can recover the stream of voice from packets arriving with the variable delay introduced by the packet-switched network.

The HPCT telephony network of Fig. 1, along with its associated service protocols, is symmetric. However, sometimes a caller may have a multimedia-capable computer and a packet-switched network connection, thereby enabling advanced services or features. However, if the callee does not have (a) a terminating multimedia computer, (b) direct access to the packet-switched network, and (c) a compatible packet-switched telephony application currently running on the terminating computer, then the call can not be completed over just a packet-switched network. In this case, a terminating circuit-switched LEC 7 supported by a terminating GC 6 may be used in the same way as in the first embodiment of the present invention, but the telephony system will have an asymmetric configuration as shown in Fig.4.

In this system, the caller's multimedia computer 4 will run a digital communications program comparable to an originating GC's protocol and therefore will serve as the originating GC from the view point of the packet-switched network 5 and the terminating GC 6, except the billing and validation of the caller may be performed by the terminating GC 6 based on the caller's access point to the packet-switched network. Similarly, where the callee has a multimedia capable computer and a packet-switched network connection but the caller does not, the telephony system of the invention may have an asymmetric configuration that is the reverse of the Fig. 4 configuration.

On the other hand, where the HPCT utilizes Gateway Computers at both ends of the packet-switched network, each GC provides a set of resources that are shared by many users and thus achieves much higher utilization in the telephony than a personal computer. Optimization of performance can be achieved by using Digital Signal Processors (DSP's) or Application Specific Integrated Circuits (ASIC's). Furthermore, the users do not have to make a large investment, operate special computer equipment and programs, or schedule calls in advance. In fact, as described relative to Figs. 5 and 7, the users may not tell any difference between using the HPCT and using their regular long-distance services, except for a much lower cost.

Considering the 50:1 compression ratio discussed above, the utilization of the circuits in circuit-switched telephony can be only 1/50 as efficient as that of the HPCT; in other words, the cost of the former can be as much as 50 times higher than that of the HPCT. Another important problem with circuit-switched telephony is the proprietary nature of the telephony switches which are the foundation of this telephony. Because switch software development is only done by the manufacturers, the cost and delay in adding new services are often frustrating and prohibiting. The HPCT, however, is based on general-purpose computers with open architecture, which can open up development and bring very cost-effective new services in a much shorter time frame.

The packet-switched network 5 of the HPCT system can be one of many types of packet-switched public data networks, such as X.25 or the emerging Asynchronous Transfer Mode (ATM) network. The ATM network is a special packet-switched network with low delay and low delay deviation, in which data is formatted into special types of packets, referred to as "cells", to achieve fast-switching. Accordingly, ATM networks are sometimes referred to as having a third type of networking, namely "cell-switched networking".

A caller can use the HPCT system as an alternative long distance telephony service ("charge service"), or the caller can use it as his/her primary long distance telephony service ("dedicated service"). Charge service can be reached from any telephone while dedicated service can be reached only from a subscriber"s dedicated telephone, such as a home phone or office phone. The alternative service is referred as "charge service" because the caller does not need to have a dedicated telephone account with the service provider; instead, the authorization is via a credit card or calling card inquiry.

To implement the charge service and the dedicated service, the invention provides two respective protocols for processing calls within the hybrid telephony system. A first protocol for the charge service is illustrated in Figs. 5 and 6. This process includes the following steps: (1) The caller first calls a local originating GC through a circuit-switched originating LEC from any telephone, and the caller's address (caller's telephone number) is relayed to the originating GC by the originating LEC. (2) The originating GC plays a voice prompt (a greeting message asking for input) to the caller asking for the callee's destination address (callee's telephone number). (3) The caller provides the address either through telephone keypad digits or through spoken digits which are recognized by the originating GC. (4) The originating GC resolves the call routing information in a manner similar to the Domain Name Service for the Internet,

obtains the packet network address (such as the IP address of the Internet) of the terminating GC, which is usually local to the callee (otherwise a toll call may be involved), and estimates the unit charge for a call going through that terminating GC. (5) The originating GC informs the caller about the charge rate, and asks for the caller's preferred payment method, such as by credit card, or through a prearranged calling card account. (6) The caller specifies the payment method either through keypad digits or through spoken digits which again are recognized by the originating GC (if this is a collect call, then the caller's spoken information about both parties is recorded and digitized to be announced later to the callee). (7) The originating GC validates the payment method through an internal or external database. (8) The originating GC sends a control message to the terminating GC, along with both party's addresses and, if the terminating GC does not know where to route the call or does not have the resource to serve the call, it responds with a negative acknowledgment and an alternative terminating GC is searched for and selected, or the caller is informed that the call can not be routed at that moment. (9) The terminating GC dials out to the callee through a circuit switched terminating LEC using the destination address it obtained from the originating GC. (10) If the call proceeds successfully through the terminating LEC, the terminating GC sends an acknowledgment back to the originating GC (or if the call proceeds unsuccessfully, such as due to busy telephone lines, the terminating GC sends this status to the originating GC in the form of a busy message). (11) The originating GC then passes the status of acknowledgment back to the caller through the originating LEC, the effect being a ring back tone (or a busy tone). (12) The callee answers the call. (13) The terminating GC passes this state change to the originating GC, which may begin billing at that time. (14) The callee starts the conversation by greeting the caller. (15) The terminating GC either receives the digitized voice data stream over a digital trunk or continuously digitizes all the voice signals over an analog trunk from the LEC which the callee is connected to, and, after possibly encrypting and compressing,
packetizes the data into packet form, the packets then being sent over the packet-switched network to the originating GC. (16) The originating GC, after possibly rearranging the packets to maintain proper packet order, unpacketizes the received data and, after possibly decompressing and decrypting, optionally converts the digitized data back to the voice signal if the connection with the LEC which the caller is connected to is analog, which is then routed to the caller over the circuit-switched network of the originating LEC. (17) The same process as described in steps 14 through 16 is performed for the caller's voice in the opposite direction.

The resulting processing in both directions supports the conversation between the two parties participating in the call.

The GC preferably provides out-of-band signaling, and the call signaling sequence for providing the charge call of Fig. 5 will now be described with reference to Fig. 6. (1) The caller's telephone number is sent to the originating GC to access a call. (2) The originating GC prompts for a destination address, such as by a dial tone. (3) The caller inputs the callee's address, such as by dialing the callee's telephone number. (4) The originating GC may provide a voice message regarding rate, and prompts for a payment method, such as by a special tone or by a voice message. (5) The caller inputs the desired method of payment, such as keypad numbers corresponding to a credit card. (6) The originating GC validates the payment method and then sends a connection request to the terminating GC. (7) The terminating GC dials the callee through the terminating LEC. (8) Depending on whether the callee's telephone is available or busy, the terminating GC sends a corresponding acknowledgment to the originating GC. (9) Depending on whether the callee's telephone is available or busy, the originating GC sends a ring back tone or a busy signal to the caller's telephone. (10) If the callee answers the telephone, an off-hook signal is sent to the terminating GC. (11) The terminating GC then sends an answer indication to the originating GC, which starts billing and sets up the in-band routing for both digitized voice data and analog voice transmission. (12) At this stage, either the callee or the caller may initiate the conversation. If initiated by the callee, the callee's telephone sends the voice greeting to the terminating GC. (13) The terminating GC either receive digitized voice data in bit stream from the terminating LEC or digitizes the analog voice, and may also perform the additional functions previously described, and then sends the digitized voice data to the originating GC. (14) The originating GC converts the voice data to analog voice by performing the functions previously described, and sends the analog voice to the caller.

A second protocol for the dedicated service on the HPCT telephony system of the invention is shown in Fig.7. This process includes the following steps: (1) The caller initiates a long-distance call by dialing a destination address (callee's telephone number) through the circuit-switched network of an LEC from his/her dedicated telephone, such as a home phone or office phone, for which a routing configuration to an originating GC is present within the LEC. The LEC routes the call to the GC, and the caller's address (caller's telephone number) is passed to the GC by the LEC, along with the destination address. (2) The originating GC authorizes the call by checking the caller's account information through an internal database or may

communicate with a centralized database for the account information, and it also resolves the call routing information using the dialed destination address. (3) The originating GC then sends a control message to a terminating GC, along with both party's addresses. If the first terminating GC does not know where to route the call or does not have the resources to service the call, it responds with a negative acknowledgment and an alternate terminating GC is searched for and selected, or the caller is informed of the negative acknowledgment. (4) The terminating GC dials out to the callee through the circuit-switched network of the terminating LEC. (5) If the call proceeds successfully through the terminating LEC, the terminating GC sends an acknowledgment back to the originating GC (the handling of the unsuccessful, most likely busy, scenario is similar to that in the first protocol). (6) The originating GC then passes the status back to the caller through the originating LEC, the effect being a ring-back tone. (7) The callee answers the call. (8) The terminating GC passes this state change to the originating GC which may start charging. (9) The callee initiates the conversation by greeting the caller. (10) The terminating GC either receive digitized voice data in bit stream from the terminating LEC or continuously digitizes all the voice signals from the callee, possibly encrypts and compresses, and packetizes the data into packets, the packets then being sent over the packet-switched network to the originating GC. (11) The originating GC, after possibly rearranging the packets to maintain proper packet order, unpacketizes the data, possibly decompresses and decrypts, and optionally converts it back to the voice signal if the connection with originating LEC is analog, which is then routed to the caller over the circuit-switched network of the originating LEC. (12) The same process described in steps 9 through 11 is performed for the caller's voice, except in the opposite direction. This processing in both directions supports the conversation between the two parties participating in the call.

While the present invention has been described in connection with a system having a circuit-switched network in the form of both a digital and an analog Local Exchange Carrier (LEC) serving analog telephone sets, it is likely that there are many instances where only the digital network interface is needed to connect to the LEC. With a digital circuit-switched network, the configuration of the corresponding GC's is simplified since it is no longer necessary for the data manipulator to convert the voice signal into digital data, and vice versa.

From the foregoing, it can be seen that the present invention provides an improved telephony system which effectively integrates voice and data in a hybrid circuit-switched and

packet-switched telephony network, while ensuring real-time high quality voice communication and calling services with low transmission and access cost. By utilizing gateway computers of telephony service providers to route calls between circuit-switched and packet-switched telephone networks, HPCT provides the benefits of packet switching to any telephone subscriber, with none of the substantial initial investments required by pure packet-switched telephony. The potential for vastly increased intelligent services due to computer based telephony, such as caller's personalized speed dialing list, callee's personalized virtual destination number, integration with electronic mails, and many others, allows for even further enhancement of the HPCT system. Furthermore, the HPCT system, which is based on general-purpose computers with open architecture, can open development of a host of new services and make them cost-effective in a much shorter time than would be required for complete conversion from conventional circuit-switched networks to entirely packet-switched networks.

While this invention has been described in the context of preferred embodiments comprising at least one circuit-switched network of an LEC, it should be clear that the principles of the invention will work equally well with other telecommunications networks and with variations of the preferred embodiment that will be apparent to those skilled in the art. Many other modifications and alternatives are possible and will occur to those skilled in the art who become familiar with the instant disclosure. Such modifications and alternatives are intended to be within the scope of the invention as defined by the claims set forth below

CLAIMS

What is claimed is:

1. A telecommunications system comprising:

an originating circuit-switched network for transmitting originating voice signals;

originating gateway means for converting said originating voice signals into packets of digital data,

terminating gateway means for converting said digital packets into terminating voice signals,

a terminating circuit-switched network for transmitting said terminating voice signals, and

a packet-switched network for transmitting said digital packets from said originating gateway means to said terminating gateway means, at least one of said originating and terminating gateway means comprising means for routing said digital packets through said packet-switched network from said

originating gateway means to said terminating gateway means.

2. A telecommunications system according to claim 1, wherein said originating gateway means comprises means for compressing said digital data, and wherein said terminating gateway means comprises means for decompressing said digital data.

3. A telecommunications system according to claim 1, wherein said originating gateway means comprises means for encrypting said digital data, and wherein said terminating gateway means comprises means for decrypting said digital data.

4. A telecommunications system according to claim 1, wherein said terminating gateway means comprises terminating buffer means for storing said digital packets prior to the conversion thereof into said terminating voice signals.

5. A telecommunications system according to claim 4, wherein said terminating gateway means further comprises means for rearranging said stored digital packets to maintain a proper packet order.

6. A telecommunications system according to claim 1, wherein said routing means provides said routing in response to dialed digits.

7. A telecommunications system according to claim 1, wherein said routing means provides said routing in response to spoken digits.

8. A telecommunications system according to claim 1, wherein said terminating circuit-switched network comprises means for transmitting first return voice signals to said terminating gateway means; wherein said terminating gateway means comprises means for converting said first return voice signals into packets of return digital data; wherein at least one of said originating and terminating gateway means comprises means for routing said return packets through said packet-switched network from said terminating gateway means to said originating gateway means; and wherein said originating gateway means comprises means for converting said return packets into second return voice signals.

9. A telecommunications system according to claim 8, wherein said originating gateway means comprises originating buffer means for storing said return packets prior to conversion thereof into said second return voice signals.

10. A telecommunications system according to claim 9, wherein said originating gateway means further comprises means for rearranging said stored return packets to maintain a proper packet order.

11. A telecommunications system comprising:

originating means for providing digital packets corresponding to originating voice signals,

terminating gateway means for converting said

digital packets into terminating voice signals,

a terminating circuit-switched network for transmitting said terminating voice signals, and

a packet-switched network for transmitting said digital packets from said originating means to said terminating gateway means, at least one of said originating means and said terminating gateway means comprising means for routing said digital packets through said packet-switched network from said originating means to said terminating gateway means.

12. A telecommunications system according to claim 11, wherein said originating means comprises means for compressing said digital data, and wherein said terminating gateway means comprises means for decompressing said digital data.

13. A telecommunications system according to claim 11, wherein said originating means comprises means for encrypting said digital data, and wherein said terminating gateway means comprises means for decrypting said digital data.

14. A telecommunications system according to claim 11, wherein said terminating gateway means comprises terminating buffer means for storing said digital packets prior to the conversion thereof into said terminating voice signals.

15. A telecommunications system according to claim 14, wherein said terminating gateway means further comprises means for rearranging said stored digital packets to maintain a proper packet order.

~~16. A telecommunications system according to claim 11,~~ wherein said routing means provides said routing in response to data received from said terminating gateway means.

17. A telecommunications system according to claim 11, wherein said routing means provides said routing in response to a typed input from a computer keyboard.

18. A telecommunications system according to claim 11, wherein said terminating circuit-switched network comprises means for transmitting first return voice signals to said terminating gateway means; wherein said terminating gateway means comprises means for converting said first return voice signals into packets of return digital data; wherein at least one of said originating means and said terminating gateway means comprises means for routing said return packets through said packet-switched network from said terminating gateway means to said originating means; and wherein said originating means comprises means for converting said return packets into second return voice signals.

19. A telecommunications system according to claim 18, wherein said originating means comprises originating buffer means for storing said return packets prior to conversion thereof into said second return voice signals.

20. A telecommunications system according to claim 19,

wherein said originating means further comprises means for rearranging said stored return packets to maintain a proper packet order.

21. A telecommunications method comprising steps of:

providing digital packets for transmission from an originating means, said digital packets corresponding to originating voice signals,

transmitting said digital packets from said originating means to a terminating gateway means through a packet-switched network, at least one of said originating means and said terminating gateway means comprising means for routing said digital packets through said packet-switched network from said originating means to said terminating gateway means,

converting said digital packets into terminating voice signals for transmission from said terminating gateway means, and

transmitting said terminating voice signals through a terminating circuit-switched network.

ABSTRACT OF THE DISCLOSURE

A telecommunication system includes an originating circuit-switched network for transmitting originating voice signals, an originating gateway computer for converting the originating voice signals into packets of digital data, a terminating gateway ~~computer for converting the digital packets into terminating~~ voice signals, a terminating circuit-switched network for transmitting the terminating voice signals, and a packet-switched network for transmitting the digital packets from the originating gateway computer to the terminating gateway computer. Either the originating or terminating gateway computer may route the digital packets through the packet-switched network from one computer to the other. At either the originating or terminating end of the system, the corresponding circuit-switched network and gateway computer may be replaced by a computer terminal running a digital communications program.